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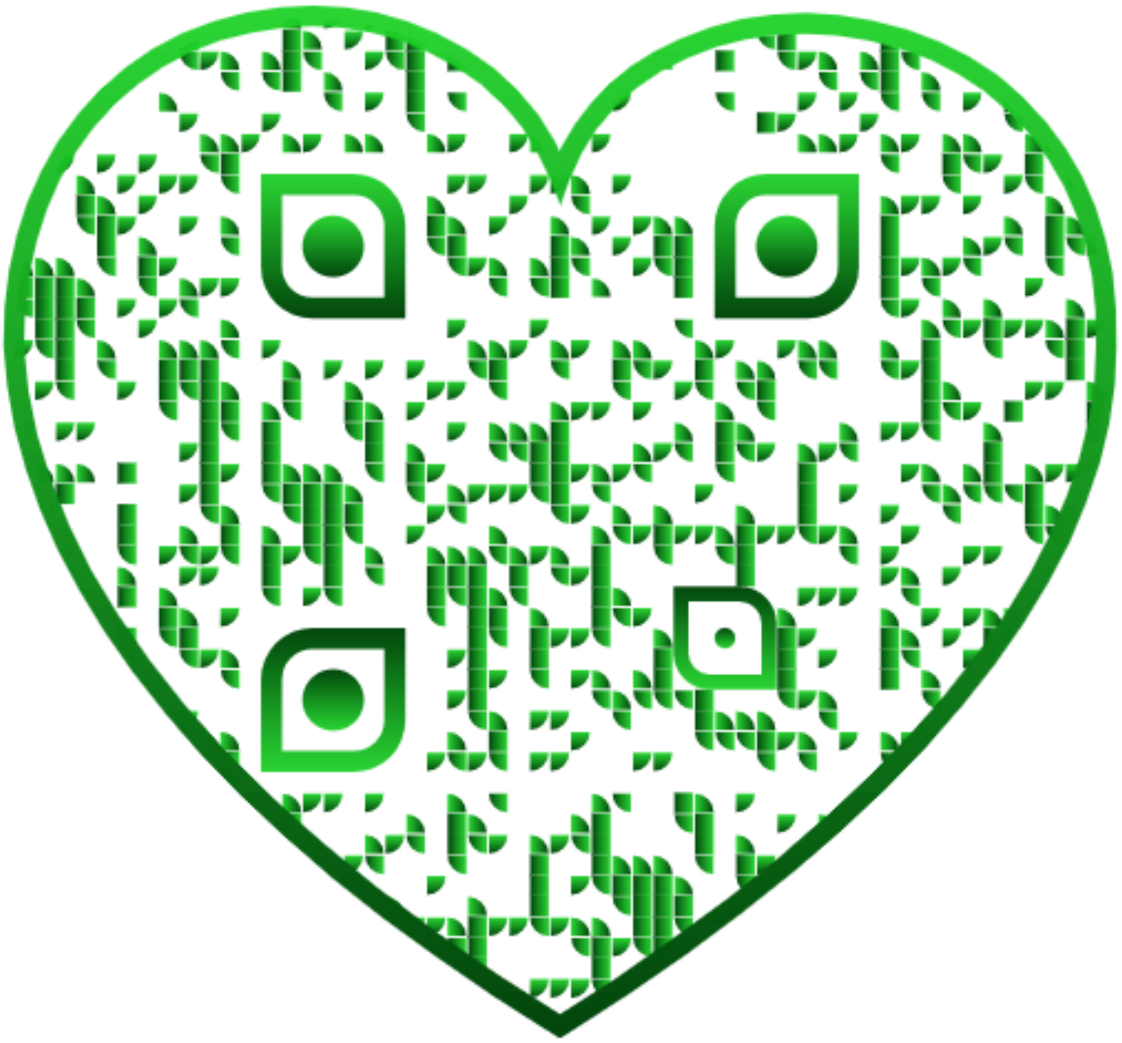

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CHANGE IS HAPPENING

Music production is a pretty insane business. Long hours, tight budgets, tough deadlines, ridiculous amounts of coffee... but for all the ups and downs navigating this wonderful and curiously creative career choice, nothing makes me more frustrated than being asked "why are there so few women?"

When I began working in this industry nearly 20 years ago, there were just a handful of us, but we have made enormous strides to readdress that imbalance since then. There is still a long way to go and there are so many of us who will never stop striving to increase diversity in all areas of the business. All too often, though, our existence has been measured through statistics. But it doesn't represent the bigger picture. I know so many engineers, producers and mixers kicking ass, making great records and winning awards. I wanted, in one place, to showcase that. No statistics, no politics, just here we are. This is what we do. Look at us all!

We have been lamenting about the "who isn't" for so long I felt it was time we celebrated the "who is" for a change... As one of the newly appointed directors of the Music Producers Guild we were able to reach out to *Sound On Sound* magazine with the idea. They loved it! And with an enthusiastic thumbs up from editor Sam Inglis, I grabbed my Rolodex (aka Instagram) and got to work.

Reaching out to about 20 people I knew in music production, I was gleefully inundated with more and more suggestions: and suddenly there were at least 50 names floating around. How do we include everyone? We wanted to maintain a majority of the format but *SOS* wanted to make changes too. Researching more diverse contributors, authors and subjects. And with more talented names being added to the list: something for the next issue!

My hope is this issue, with its broad readership, will begin to create real visibility for those killing it in the industry. To inspire those starting out, or contemplating getting into music production, and to help the decision makers see what a wonderfully diverse pool of talented producers and engineers there are. You can't be what you can't see! I particularly wanted to pay tribute to those who I looked up to when I first started. Linda Perry, Sylvia Massey, Emily Lazar, Ann Mincieli. Those with experience who have been doing this a while and have paved the way for a new generation of superstars.

It's been a long time coming but change is happening; perception just needs to catch up.

Catherine Marks

"It's been a long time coming but change is happening; perception just needs to catch up."

Photo: Phoebe Fox

Guest Editor Catherine Marks is Executive Director of the Music Producers Guild and a Grammy Award-winning producer and mix engineer, with credits including Wolf Alice, Foals, St Vincent and boygenius.

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Slate Digital Transient Shaper

Transient Processing Plug-in

Slate's latest plug-in offers precise control over your transients.

MATT HOUGHTON

As with other plug-ins for Slate Digital's Virtual Mix Rack, Transient Shaper is available in the usual plug-in formats for macOS and Windows, and is licensed using the iLok system (though a physical iLok isn't required). There are now plenty of transient processor plug-ins available, but this one includes some useful features that make it very easy to target the aspects of a sound you wish to process with precision.

Snap To It!

At the top of the 500-series-style GUI are controls to boost/cut the transient portions of a signal. In the middle there's a similar set of controls to manipulate the sustain, and a bottom section hosts global functions: a very handy 'listen' facility, a stereo link toggle, a make-up gain control plus protective clipper (very useful, given the gain in which this sort of processing can result), and a wet/dry Mix control.

The Transient and Sustain sections not only allow you to boost/cut (on a scale of -10 to +10) but also have a side-chain bracket filter (a HPF plus a LPF) with which you can target the processing at a specific frequency range. There are no frequency markings on this slider but when you drag a filter handle the frequency is displayed numerically — a good way to ensure the GUI remains uncluttered. (A tiny misgiving is that it's fairly easy to grab the GUI near to a slider, and dragging that instead will reorder the modules in Virtual Mix Rack.) These two sections also feature a meter that indicates how much boost/cut is being applied by each section, and the Transient section also features a Warmth saturation control, which can serve to tame any harshness in the transients.

The Listen button allows you to hear your choice of the processed output, only the transient portion of the signal, or only the sustain portion. And the Time knob adjusts the 'time crossover' that determines what's treated as a transient and what as sustain. The two used in combination make it incredibly easy to hear what you're doing and thus to target your processing with laser-like precision. I've found that an effective tactic can be to solo the Transient section, then adjust the Time until you hear exactly what it is you want to boost/cut, before homing in on the specifics of the sound using the filters. It's trivially easy to establish the optimum settings that way.

In Use

Transient processors are always great for tailoring individual drums, but when deployed on more complex loops or the stereo drum bus they can be blunt tools. The facilities on offer here make this one less limited in that respect. For instance, I was able to tame a flabby kick in a stereo drum loop with very little collateral damage to the snare, and the tightened bottom end meant I was able to use VMR's SSL-style EQ to apply a generous LF boost and notch out an annoying 'knock' around 180-200 Hz. None of that would have been possible had I not first addressed the 'flab'. Similarly, I could bring some long cymbal tails in the drum bus under control, by bringing down the sustain only in the relevant frequencies, before restoring some top-end 'life' using Revival (a free module for Virtual Mix Rack).

It's not just for drums, though. I had great success targeting the 'spikiness' in a fingerpicked acoustic guitar part, which sounded good in isolation but distracted from the vocal. I was also able to bring an annoying 'twang' under control on an over-enthusiastically picked electric bass.

I love transient designers, and use them a lot in mixing. This one offers much more precise control than most I've used, in no small part due to the monitoring and timing controls. It makes sculpting a sound you like very quick, easy and satisfying.

Features like the transient saturation and safety clipper are really thoughtful too, and, importantly, if you already use Slate's Virtual Mix Rack this module could help you get a lot more mileage out of the various EQ and enhancement plug-ins you already have in that environment.

summary

A transient manipulation plug-in that manages to keep things simple and intuitive while offering much more precision than most — a welcome addition to the Virtual Mix Rack environment.

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Toontrack Soul Roads EKX

EZkeys Expansion

Hot on the heels of EZkeys 2 comes its first sound expansion, Soul Roads EKX.

JOHN WALDEN

Toontrack have wasted no time with the first sound expansion option for EZkeys 2. There are no prizes for guessing the target of Soul Roads — it's the classic sound of the Rhodes electric piano — but what we get here is in fact two examples: an early MkI from 1970 and a MkII from the 1980s. Within EZkeys itself, these appear as two separate instruments, each with its own collection of presets. Toontrack already offered their Classic Electrics expansion for EZkeys 1 and that included a very usable Rhodes MkI (alongside a Wurlitzer 200A), so just what does Soul Roads bring to the party?

The Soul Enchilada

Sonically, both of the new pianos sound impressive and, as with the actual MkI and MkII, while they have a similar character, there are some obvious tonal differences. For those trying to get as close as they can to the sound of a specific musical era — '70s or '80s — the sonic details will be important. For everyone else, it's simply two alternative electric piano flavours to choose between. It's win-win though, as they both sound beautiful. Whether you want old-school soul, funk, pop ballads or some R&B (traditional or modern!), there are great options here.

The supplied presets provide plenty of variations for both core instruments. As well as a Basic option for both, there are presets featuring various ambience options such as tape delay or plate reverb, a range of modulation styles including chorus and phaser, rotary speakers, various amplified sounds (included the excellent 'Panned Amps' preset), an interesting 'Talking Piano' preset (which features a very cool envelope filter effect) and 'Filtered Vinyl' presets with tape flutter and vinyl noise that can be dialled in for a suitably lo-fi sound.

Unsurprisingly, on a technical front, the Soul Roads is somewhat larger than its EZkeys 1 equivalent, clocking in at 1.2GB, and most likely representing a more sophisticated and/or detailed approach to the sampling. That said, individual presets from either expansion pack, once loaded into EZkeys 2, have similar RAM needs (around the 400MB size). There are obvious sound differences, though; compared side by side, while Classic Electrics still sounds good, the Soul Roads pianos just seem to have a greater depth and a much more satisfying tonal response to playing dynamics. And, designed for the EZkeys 2 engine, with its Effects panel on the Keys page, Soul Roads presets generally give you more options for adjusting elements of the sound.

The EKX is supplied with a suitable soul-inspired MIDI library and, if you need more, alongside the EKX MIDI content, Toontrack have also released two further MIDI expansion packs — Retro Soul and Modern Soul — both of which contain plenty of excellent performance ideas. The Soul Roads sounds themselves will, of course, also suit lots of other musical styles represented within the EZkeys MIDI expansion options.

Conclusion

Classic electric piano sounds were an obvious first candidate for Toontrack on the back of the EZkeys 2 release and Soul Roads captures these sounds brilliantly. If soul, funk, blues, ballads, R&B or chilled pop are your thing, Soul Roads is going to tick lots of sonic boxes.

summary

Soul Roads delivers two generations of Rhodes electric pianos alongside some excellent MIDI content. The first EKX for EZkeys 2 sounds great.

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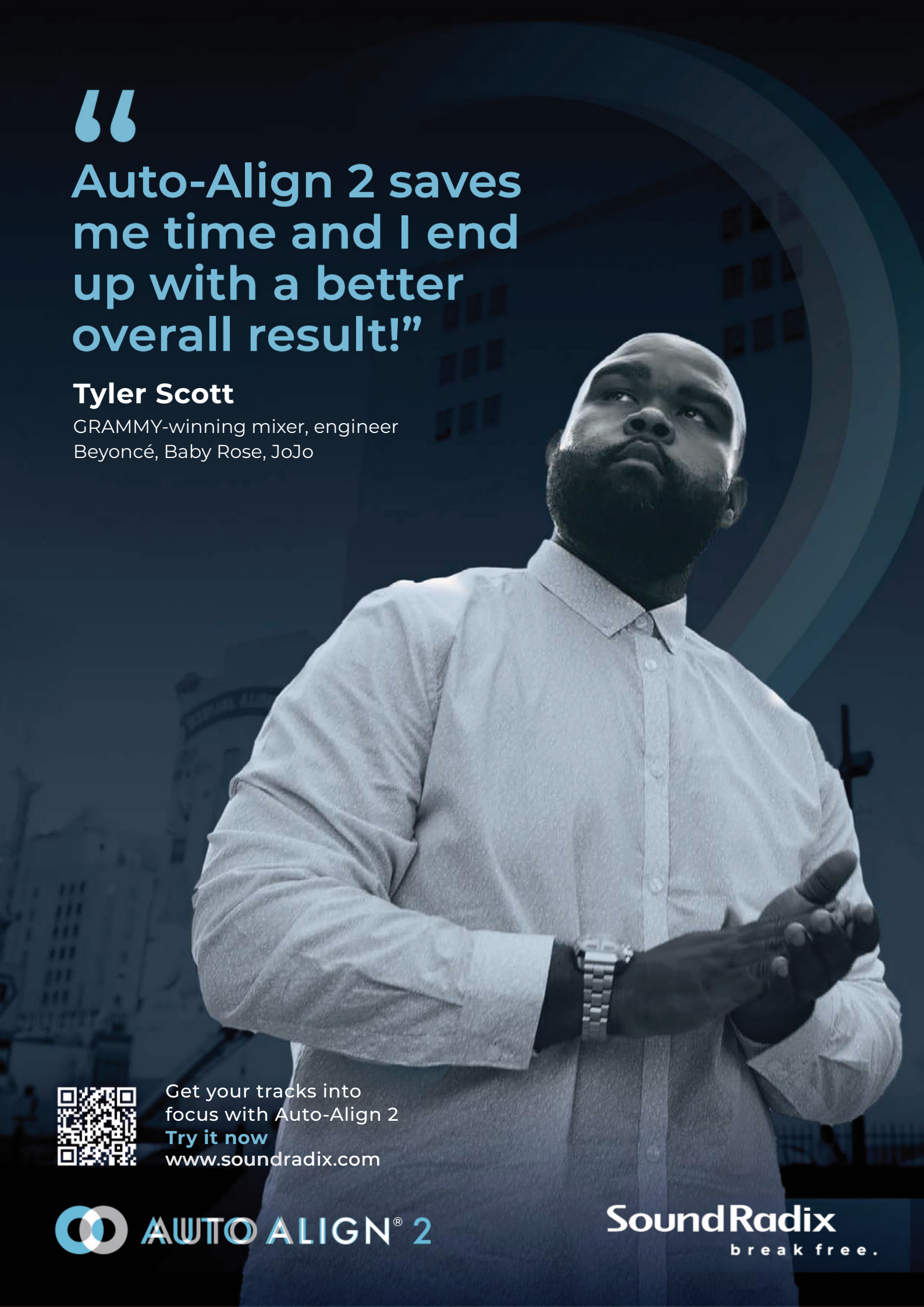
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Loudspeaker Design Cookbook

Book Review

In print for almost half a century, the eighth edition of Vance Dickason's loudspeaker bible is as relevant as ever.

PHIL WARD

Vance Dickason's *Loudspeaker Design Cookbook* (ISBN 9798549209930) is as much an institution as it is simply a dense volume on the theory and practice of speaker design. I say that not only because, with its eighth edition, it's been around now for 46 years, but because it is and always has been truly unique. There are quite a few books on speaker design out there, but no other is quite like Vance Dickason's.

Along with his very long-term commitment to writing about speaker design and engineering, Vance has a parallel career as a speaker design consultant working across pretty much every sector where speakers are present — from simple consumer electronics products, through in-car audio to the highest of high-end hi-fi. He's also the editor of *Voice Coil*, the loudspeaker designer's trade magazine (I bet you didn't know we speaker geeks have our own monthly magazine). Vance, perhaps as much as anybody else in the speaker design world, has earned the right to author the definitive book on the subject.

While I'd wager that the majority of *Loudspeaker Design Cookbook* readers fall into the 'amateur speaker-building hobbyist' category and probably live in the US (where speaker building as a hobby is rather more popular than it is in the UK), I'd also wager that if you were to glance along the reference bookshelf in pretty much any professional speaker R&D office you'd find a copy of *LDC*. And the reason for that goes back to to a word I used up in the first sentence: dense. If you could rank technical books in terms of nuggets of information per paragraph, *LDC* would be up near the top of the charts.

Nuts & Bolts

Beginning, as it does, with chapters on subjects that cover the deep nuts and bolts of basic loudspeaker engineering, electro-acoustic theory, enclosure construction and crossover design, *LDC* at first glance looks perhaps a little too niche for the typical *Sound On Sound* reader. But for the eighth edition, along with very significant revisions and expansions of existing material, and a brand-new chapter on computer-based speaker design tools, Vance has added a couple of chapters towards the end of the book that cover room acoustics and room correction systems, and the history, engineering and application of studio monitors. For those of us for whom these subjects are of day-job significance, I think these two chapters are worth the *LDC* purchase price alone. And once you've got used to Vance's straightforward yet engaging writing style, one that manages to make approachable even the most complex electro-acoustic concepts, I reckon the rest of the book will begin to draw you in. You'll become a speaker geek by stealth.

LDC, is not a book you'll read from cover to cover, and nobody's ever going to write that it's 'unputdownable', but as a resource to dip in and out of and which contains a very significant proportion of everything there is to know about the practice of speaker design, it is unique and completely unrivalled.

summary

Comprehensive and engaging, the *Loudspeaker Design Cookbook* is essential reading for any budding speaker designer.

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SSL G3 MultiBusComp

SAM INGLIS

Multiband Compressor Plug-in

Solid State Logic have added some new strings to their G-series compressor's bow.

How can you improve upon the classic SSL G-series bus compressor? Lots of developers have taken it upon themselves to provide answers, adding extra features such as additional ratio settings, side-chain filtering options and wet/dry mix controls. SSL's own Bus+ hardware unit takes this trend to its logical extreme in hardware; now, their G3 MultiBusComp plug-in does the same in software.

The Big G

Available for macOS and Windows in all the usual native formats, and authorised using iLok, G3 MultiBusComp is available to buy outright, through a rent-to-own scheme or as part of the SSL Complete subscription package. It incorporates all of the above-mentioned bonus features and more. But, as the name suggests, it isn't simply a software recreation of the G-series bus compressor. It's actually three G-series compressors, operating separately on the low, mid and high-frequency bands.

That is, if you choose to set it up like that. Adjacent bands can be 'linked', which is a slightly confusing usage; in practice, 'linking' the upper and middle bands, for example, simply disables the upper band and its crossover, leaving you with a two-band compressor. And the crossovers themselves can be set anywhere from 20Hz to 20kHz, so there's nothing to stop you having two bass bands and a single mid/high band, or *vice versa*.

Each band naturally has its own time constant, ratio and threshold settings, as well as make-up gain. To the original 2:1, 4:1 and 10:1 ratios, this plug-in adds gentler 1.5:1 and intermediate 3:1 and 5:1 options, and wet/dry mix is a per-band setting, as is the choice of internal or external side-chain. Each band also has a button with a headphones icon that lets you audition its filtered side-chain signal. Additionally, you can introduce variable levels of saturation modelled on the original G-series compressor circuit, again on a per-band basis.

Visual feedback is likewise presented separately for each band, with VU-style gain reduction meters at the top and an animated bargraph at the bottom. The crossover points and thresholds are represented on the latter as vertical and horizontal lines respectively, which can be dragged up and down with the mouse. Cick the View: BANDS icon to the left of »

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Engaging the side-chain filters for a band effectively uncouples the side-chain EQ from the crossovers in the audio path, in this case to allow compression in the mid band to be triggered by low-frequency peaks.

this area, and it switches to show the side-chain EQ page for each band, where your choice of low, high and bell filters can be applied. The frequencies of each are fully variable, and the Q of the bell filter is supposed to be adjustable using the mouse scroll wheel, though I couldn't make that work using my Apple Magic Mouse.

Throw in global input and output level controls and a global wet/dry mix control, and that's G3 MultiBusComp in a nutshell. Everything fits nicely into a single window, and apart from the naming of the Link controls, everything should be pretty much self-evident to anyone who's used a plug-in compressor before. Having said that, one facility I did miss is the ability to actually link controls across bands, such that, for example, all of the thresholds could be moved at once.

Powers Of Three

A standard G-series compressor is a pretty powerful beast. Used wisely, it can add that elusive 'glue'. Used with a certain amount of abandon, it can inject an otherwise pedestrian mix with real rock & roll energy. And, used unwisely, it can send everything to rack and ruin. G3 MultiBusComp offers many additional ways of achieving all three possible results, and more!

First of all, of course, it can literally act as a standard G-series compressor. If you link all three bands, what you have is in effect a full-range implementation of the G-series design, with extras such as side-chain filtering and the additional ratio settings. Just for a laugh, I thought I would attempt a null test between G3 MultiBusComp and SSL's single-band Bus Compressor 2 plug-in; I couldn't quite get them to cancel completely, but they can certainly sound very similar.

Moving on to relatively subtle applications for G3's multiband processing, one of the main reasons for adding side-chain EQ to a G-series bus compressor design is to make it easier to tailor the way the compressor responds to low frequencies. G3 offers significantly more control in this respect, and what I really like is that once you enable the high and low side-chain filters, they operate independently of the crossovers in the audio path. This means that, for example, you can have compression in the mid band triggered by low frequencies, which could be a neat way of achieving the French house pumping effect; or you could have a narrow mid band triggered by a much broader side-chain signal, for example to tame a vocal resonance that occurs only on loud notes. If you want a punchy midrange with a controlled, tight low end, meanwhile, that's easy to configure, without losing the characteristic aggression of the G-series sound.

Turn down the threshold or crank up the ratio controls, and G3 MultiBusComp can exert a pretty firm hand on the signal. Such settings are less usable on the mix bus, unless you like rack and ruin, but they can be highly effective on individual sources. One project I was working on had a DI'ed bass guitar that was flabby and lacked definition; a combination of firm control below 200Hz and aggressive push in the midrange sorted it out nicely, and the saturation — which is relatively subtle — added the finishing touch. G3 MultiBusComp is also a great tool for working with vocals: by setting the controls judiciously you can simultaneously tame proximity effect, dial in midrange emphasis and compression for clarity and use the top band as a de-esser.

Banding Together

By and large, multiband dynamics processors tend to be clean-sounding devices optimised for problem-solving and subtle tonal correction. G3 MultiBusComp is perfectly capable of operating in stealth mode if you want it to, but you wouldn't call it surgical, and there would perhaps be more versatile choices if that's all you are looking for. Its real strength is to extend the assertive dynamic control we associate with the G-series bus compressor design into new realms. It's quite often the case that the desirable artefacts of compression, on the mix bus and elsewhere, are spoiled by unwanted consequences such as a loss of impact in the low end. G3 MultiBusComp makes it easier to implement compression as an effect without the unwanted side-effects.

summary

SSL's multiband implementation of the G-series compressor is well thought-out and genuinely extends its usefulness.

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GForce M-Tron Pro IV

Software Instrument

GForce's software Mellotron just keeps getting better.

GORDON REID

So, you already manufacture what many feel to be the pre-eminent Mellotron emulation, yet you feel that it's time to update and extend it, offering your customers more than before and, let's be honest, making a few extra shekels at the same time. But how? Happily, GForce's developers must have found answers that they liked because I now have in front of me two M-Trons — the original M-Tron Pro running on an Intel Mac, and the latest M-Tron Pro IV — now compatible with Apple Silicon — running on an M1 Pro. (Don't worry, it's PC-compatible too.)

New Tricks

The new version looks very different from the original, and there's no doubt that its scalable GUI is an improvement. The Mellotron-y bits are more Mellotron-y, and the modern features look and feel more, umm... modern-y. Sure, many of its capabilities are the same as before, but a closer look shows that there are several updates and additions. The new wow and flutter controls allow you to age the instrument in ways that we hated in the 1970s, the LFO has been augmented, there's a second filter family, there's a second aftertouch destination, the key range has been extended by seven semitones, and the effects section has been enhanced to include an amp emulator, an improved ensemble and a reverb section. Despite all of these additions, I find the GUI and its associated patch manager to be clearer and easier to use than before. What's more, the M-Tron Pro IV no longer suffers from the fake coffee stains of previous versions. What's not to like?

A Mellotron is at heart a sample player so, as well as its enhanced synthesis and effects, the M-Tron Pro IV has an enormous underlying library. This ranges from instantly recognisable Mellotron MkII and M400 strings, brasses, flutes and choirs, through numerous MkII rhythm tapes and M300 tapes, many lesser-known Mellotron sounds, Birotrons, Chamberlins and, perhaps of most interest to aficionados, obscuria rescued from custom tapes recorded for bands as diverse as Black Sabbath and Tangerine Dream. The review copy was also supplied with four free expansion packs: ChamberTron Extras, Compute-a-Tune, Piano Bass 2 and Isolation Choir 2020, all of which include interesting and potentially useful sounds.

Obviously, none of this would matter if the results were bland or uninspiring, but the sound quality is great, the Mellotron's character has been maintained even after tidying up some of the more wayward notes on the original tapes, and looped versions of many sounds are available in case you want to sidestep the Mellotron's eight-second limitation.

Conclusion

All of this will be of little interest if your desire for Mellotron sounds is confined to the 'Strawberry Fields' flutes, the three violins that almost single-handedly defined prog rock, and dead men singing. There are many alternatives for these and, while the M-Tron Pro IV and its predecessors sound more authentic, I've been using samplers and workstations to play these sounds on stage for more than three decades. 'Careful With That Axe Eugene' played on a Roland XP-80? It sounds like sacrilege, but you would be surprised! However, if your ambitions extend beyond the more famous Mellotron sounds, M-Tron Pro IV's huge library and updated synthesis/effects engine allows you to go much further. Perhaps it doesn't have the heft and authority of my Mellotron MkII (or indeed, GForce's own M-Tron MkII soft synth), but I suspect that it's still the most widely useful recreation of almost all things Mellotron-y, and now even more so.

summary

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PSP auralComp

Immersive Audio Compressor Plug-in

Intuitive and powerful, this processor can handle anything from mono up to 7.1.4 Atmos sources.

TREVOR MICHAEL

AuralComp isn't PSP Audioware's first plug-in to support Atmos — that honour goes to auralControl, which was released last year and enables you to rebalance the levels in multi-channel tracks — but PSP made their name with some great-sounding effects and processors for music, and this is their first signal processor intended for immersive audio. As a multi-channel compressor and brickwall limiter, auralComp is a broadly similar sort of tool to Waves Spherix, which I reviewed back in *SOS* February 2023: www.soundonsound.com/reviews/waves-spherix. But a key difference, and an improvement in my opinion, is that auralComp isn't limited to use with 7.1.2 or 7.1.4 channel widths: you can use it on mono, stereo sources (and can process stereo in M-S), quad, 5.0... in fact, any number of channels, all the way up to 7.1.4. auralComp supports the AAX, VST3 and AU plug-in formats for DAWs running on macOS or Windows, and can be authorised on up to three (of the licence holder's) computers.

Overview

To develop auralComp, PSP partnered with Dutch engineer and producer Ronald Prent, who is known for his work with Simple Minds, Depeche Mode, Def Leppard, Tina Turner, Rammstein and the Scorpions, and many more. Ronald has specialised in surround sound mixing since the early '00s, at which time he helped with the development and testing for Sony's SACD format and worked with API to create a multi-channel version of their 2500 compressor. With the recent boom in immersive audio production in mind, he reached out to PSP to see if they'd be interested in developing a plug-in in built around the same concepts, but taking the ideas further where digital technology allows. (Having said that, I should make clear that this is not a model of API's compressor!)

On opening the plug-in, each channel's compressor is displayed in a separate 'strip' and, as you can see on the main screenshot, the user-adjustable parameters are the sort of thing you'd expect to find on any compressor, with knobs for attack (0.1ms to 160ms), release (25ms to 2500ms), threshold (-40dB to +10dB), make-up gain (variable up to +20dB), ratio (1:1 to infinity:1), and wet/dry mix. There's also a group of three LED-style bar meters. These indicate the input and output levels and, in between, the amount of gain reduction being applied. There's no labelling or current/maximum readout to confirm what's going on, which seems an oversight — uncertain about what value each LED segment represented, I sometimes found it tricky to tell exactly how much gain reduction was being applied. Having said that, the layout makes it easy to match the output level to the input level, and the make-up gain you apply to do that gives you some sort of indication of the gain reduction taking place.

On the right-hand side is a 'quick select' control section that makes it super simple to dial in the desired amount of compression on whichever channels you wish to process. Channels can be grouped so that, for example, you can adjust the threshold of all three front channels simultaneously, while leaving the thresholds of the other channels alone. The available groups are All, Fronts, Sides, Rears and Tops, and because each group is colour-coded it's always clear what you're about to change when you grab a control. At the bottom of each strip is the option to switch the compressor in/out for that channel.

Chain Gang

Some might find the compressor's side-chain 'circuit' fairly complicated, but it's useful and extremely well thought out, and the possibilities are myriad. There's the link control on each channel, perhaps inspired by Prent's experience with the API 2500, and when this is set to zero percent, the compressor reacts only to the channel's own input signal. As you turn the knob towards 100 it becomes increasingly sensitive to the selected side-chain signal, which is where things start to get interesting... Each channel's side-chain can be your choice of eight submixes of all the channels. By default, these are called Alpha, Beta... and so on up to Theta, but they can be renamed. You select the desired submix »

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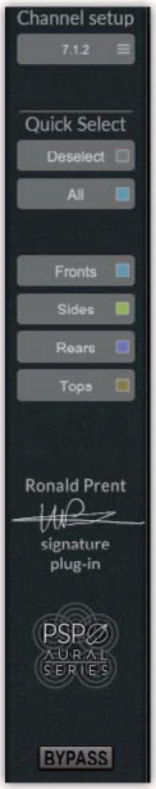


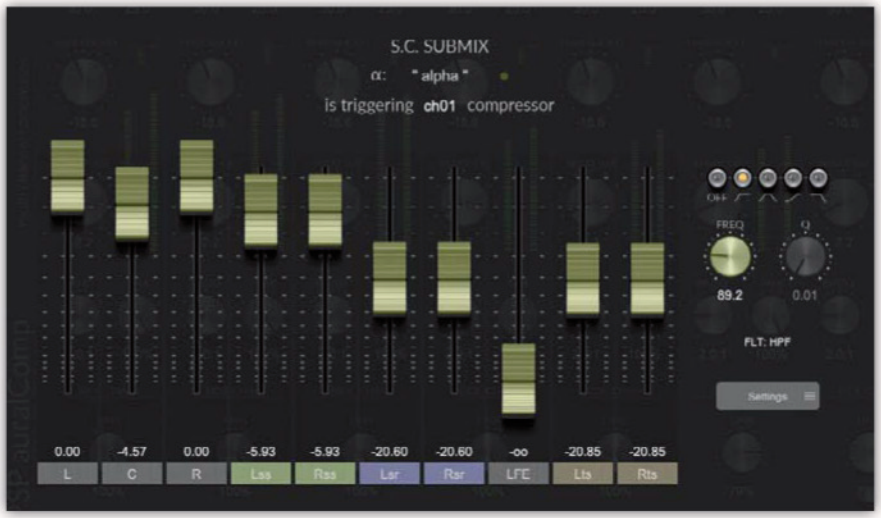
The channel grouping facilities make working with multi-channel formats quick and easy.

at the bottom of the strip, and below that is a helpful side-chain display window, which shows you a visual representation of the level of each channel in the submix, and how it corresponds to the actual side-chain level triggering the compression.

Importantly, you can tweak the side-chain submixes: clicking in the top right of the side-chain display window opens up a submixer with which you can adjust the levels in each submix — this can be done very quickly thanks to the quick select group controls I mentioned above. There are also a few filters available. Options are an HPF, adjustable from 20Hz to 20kHz; a band-pass filter, again adjustable from 20Hz to 20kHz and with an adjustable Q (0.01 to 10); an Equal Power filter (a curve designed by PSP that they say is intended to equalise energy across the entire spectrum); and a 20Hz to 20kHz LPF.

The final thing to mention about the compressor is the side-chain summing option, which applies to all the side-chain submixes. There are three choices: Natural is a conventionally summed submix, while Rectified rectifies the audio signal before summing, and when RMS is selected an averaged signal is used to trigger compression. PSP tell me that there are no rules — that the best choice is really a matter of taste — but Ronald Prent finds that RMS works well for him on classical recordings and normally prefers Rectified when working with rock. Personally I usually tended to prefer the Natural setting, but I'm sure plenty of people will find this subtle option for shaping how the compressor reacts to the audio signal useful.

In use, I found auralComp's compressor to sound big, warm, and open. On the main Dolby Atmos 7.1.2 bed for the project I was working on at the time, a little gentle compression really helped to 'finish



auralComp boasts a sophisticated internal side-chain routing system that allows any channel or group to respond to a custom mix of the other channels.

off' the mix — and I even found myself wishing I'd had access to this plug-in a couple of weeks earlier, for a project I'd recently finished. It was also easy to 'crunch up' multi-channel drum busses and then use the mix control to dial in the amount of compression to taste. It can go from gentle to punchy, to crunchy. In the manual, PSP suggest that "scenarios like 'rears are listening only to the fronts, sides are listening partially to the centre channel, ceilings are listening to the high-end information of all channels, and the LFE is excluded' are very simple to set up." I couldn't agree more: the degree of control offered in the side-chain section really does give you endless possibilities. Being very picky, I'd love to have an external side-chain option too, so that other instances of auralComp could be applied across individual objects and multi-channel busses and be keyed from the same source material. Perhaps that's something to consider for a future update.

On The Level

To the right of the compressor channels is another strip, which hosts global ±10dB input and output level controls and, tucked in between these, a switchable brickwall limiter that works independently of the compressor side-chain options. There's nothing fancy in terms of control here, but that's all right by me. Basically, you set the desired ceiling level (-20 to 0 dBFS) and the signal never exceeds that on any channel. If you want things louder, turn up and hit the limiter harder. There's also a Drive control, adjustable from 0 to +20 dB. As a visual indication of where you've set the ceiling, a helpful blue line appears on all the channels' output LED strips. To my ears, the limiter sounds really smooth, and can be pushed pretty hard if desired. I found across my Atmos bed I could crank the Drive and hear the mix starting to change, with the ambience and reverb in the rear channels coming up in volume, long before my ears could detect any undesirable artefacts.

My overall feeling? I love this plug-in — from now on, it will definitely be a staple on my Dolby Atmos mix sessions. My biggest gripe is with the metering (more information would be welcome) but that wouldn't stop me using it, and who knows, maybe that can be improved on in future versions. External side-chain support would be another welcome addition. But it would be unfair to focus on what this plug-in doesn't do, because what it does do, it does it very well indeed. auralComp's compressor is very pleasing to the ear and very quick to set up, and the limiter super-easy to use and transparent sounding. Being able to use this processor on a variety of channel widths is a real plus, too, because not every mix will be an Atmos one. PSP offer a free 30-day trial, and if you're working in immersive audio I'd say that this is definitely a plug-in you should take for a spin.

summary

A thoughtfully designed compressor that offers various side-chain routing options for immersive audio — but can still deliver the goods with good old-fashioned mono and stereo!

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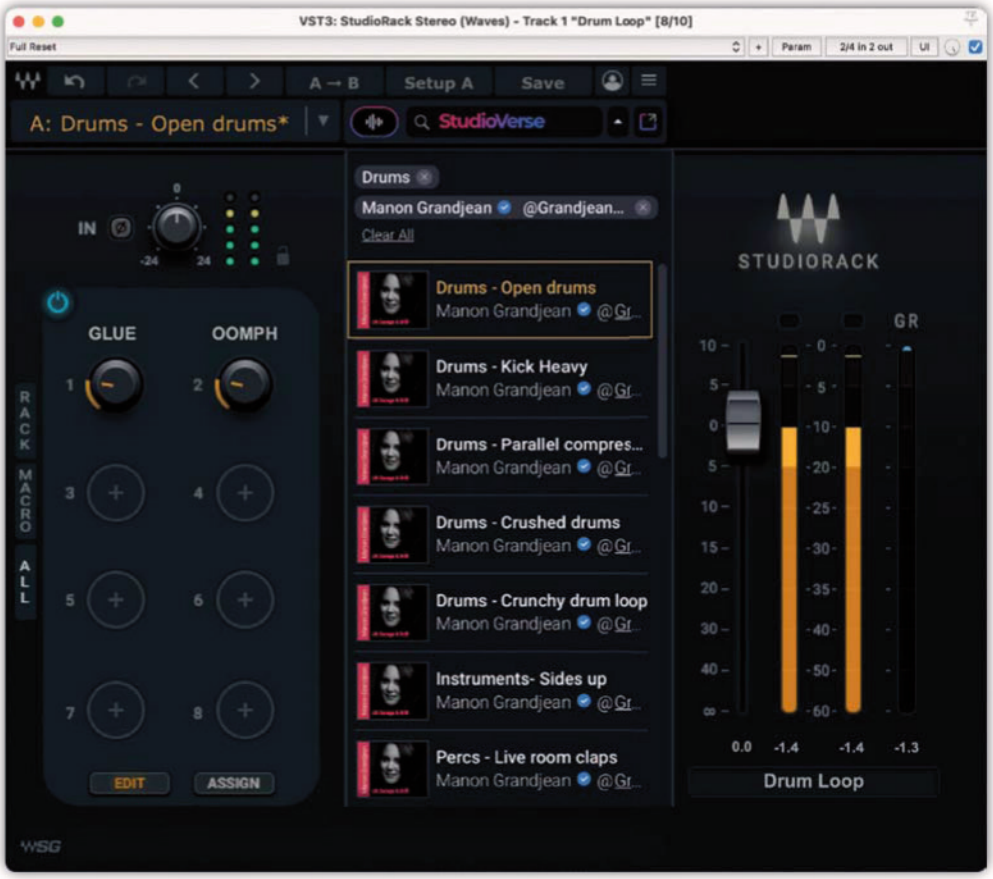

Plug-in Chainer & Preset System

This clever solution for combining processors is much more than just a plug-in host...

MATT HOUGHTON

I'll leave debating the pros and cons of plug-in subscriptions for another day, but I think it's fair to say that, for very little upfront cost, Waves' Essential and Ultimate subscription bundles offer access to a vast array of plug-in processors and effects. Choice can be a good thing, of course, but novice mixers can also find such bundles overwhelming: just where do you start? Well, one possible answer is StudioVerse.

Chapter & Verse

This new AI-powered signal-chain preset library is embedded in their free StudioRack, launched in 2022. This plug-in sits in a DAW insert slot but is itself a plug-in host with multiple insert slots, and allows you to load, create and save full processing chains that, thanks to the included Parallel Split and MultiBand Split plug-ins, can be pretty sophisticated. Crucially, StudioRack has eight assignable macro knobs that can control any parameter(s) in the chain (which, since Waves V14, can include third-party plug-ins). You can specify the control range and restrict and/or invert the parameter range too, so you could, for example, use one control to raise the signal level coming into the chain while bringing down the output. Naturally, some presets make far more imaginative use of this! MIDI Learn also makes it a breeze to map hardware controllers to macro knobs.

StudioRack always had presets but Waves have now added a StudioVerse search bar that directly accesses an online library of presets, almost 3000 of which have been created by the artists and engineers with whom Waves have developed a relationship over the years. You'll find chains from the likes of Lu Diaz, Manon Grandjean, Chris Lord-Alge, Andrew Schepps and more, and there are artist-based presets too (eg. a Beyoncé vocal chain). Being an online library, I'm sure this will continue to grow, and users can also upload and share the presets they create.

As I said, too much choice can be problematic, but a tag-based filter makes it super easy to locate suitable presets. Enter a term in the search bar and you're offered a selection of tags to restrict the search. Enter 'Drums', for instance, and you'll see tags for Drums, Acoustic Drums, Electronic Drums, Sampled Drums and Synth Drums. Choose one, then type 'Rock' and you can limit that drum search to Rock, Hard Rock or Progressive Rock. You can also filter for only presets using the plug-ins you own (it will work if you own a collection of Waves plug-ins, not just the subscriptions). The presets appear in a list below the bar, and as well as the preset name each entry provides details of who created it, along with a picture and colour-coding to make it easy to see creators you like/trust. Click a preset and, after a brief delay the preset is loaded. Click the creator instead, and your browser takes you to their entry on the StudioVerse website, where you can browse their other chains, or explore the StudioVerse more generally. You can copy/paste or drag/drop presets from the website directly into StudioRack too, which is a nice touch.

The AI side of things offers an alternative approach. Hit the waveform button by the search bar then hit play on your DAW. After a few seconds assessing your audio, StudioVerse offers a menu of presets relating to the source. For example, with StudioRack inserted on a bass guitar track, the algorithm recognised it as a bass guitar, and automatically applied the Bass Guitar tag to give me a menu of relevant presets. It's a neat idea and generally works well on individual parts or instrument busses, but seems less capable with »

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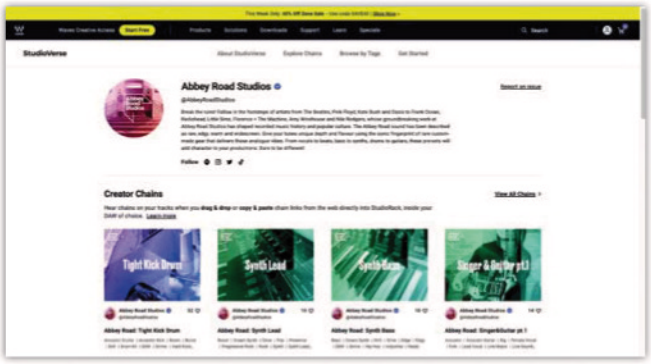
» more complex material: playing a full mix, I wasn't greeted with the mastering presets I'd hoped for. Being based on machine learning, though, there's scope to update this part of the equation over time.

The Pros & Cons Of Presets

I have to admit to approaching this review with a healthy dollop of scepticism. The thing you need to bear in mind about presets is that what you need a plug-in or signal chain to do will always vary according to the signal you're feeding into it, so it makes little sense for some parameters to be pre-determined. Always 'boosting by 2dB at 50Hz' for example, makes little sense; sometimes there's too much at 50Hz already. But it may well be helpful to have a preset with EQ bands set to certain filter types, frequencies and bandwidths — if it's an appropriate choice, just pulling the gains up/down will often get you a long way very swiftly.

Despite my initial apprehension, I enjoyed my journey to the centre of the StudioVerse, and found plenty to challenge my assumptions. Inevitably, many presets weren't to my personal taste, but whittling them down to find something I was happy with was very quick and easy. More importantly, when you load a preset, you have the key parameters laid out in StudioRack's macro section: not only does this mean you don't have to open/close multiple plug-ins, but it also prevents you going down a time-wasting rabbit-hole tweaking every parameter of the underlying plug-ins. Purely as an aid to keeping your focus on what matters, some users could find StudioVerse invaluable. You *can* dig deeper if you wish, though. You can see the signal chain laid out and access/bypass individual plug-ins just as if you were in your DAW. So you can easily refine 'nearly but not quite' presets — I often found that I wanted to dial down the grit, wow and flutter in some of the more 'vintage' effects, or to tame the reverb and delay — and save the result for next time.

Speaking of digging deeper, a whole other side to this thing is learning. One of the most fascinating things was seeing the different ways some engineers approached the same objectives: not only can you see which plug-ins they decided to use, but also in what sequence, with what sections switched on/off, and which parameters they chose to map to the macros to achieve a certain effect. The macro names often give you a sense of their intention too: to give drums 'oomph', for instance, Manon Grandjean decided one macro would control the Intensity parameter of Waves Rbass, inserted only on the low frequencies, and with the polarity inverted. Similarly, you can figure out different engineers' ways of making backing vocals sound wider, and so on. It's not always *completely* obvious why certain choices have been made, but a willing student could learn a hell of a lot, and often faster than if they were to sit through hours of YouTube!



The StudioVerse website offers another way to discover presets, which can be copied/pasted or drag-and-dropped to an instance of StudioRack in your DAW.

Bottom Line

It's not hard to see the appeal for those who are less technically inclined and prefer to stick to presets; you can access some very cool effect chains very quickly, and with all the key controls in one place, they're pretty easy to tweak. But StudioRack could also be helpful for more seasoned engineers wanting to keep their focus on the music and creative flow, rather than get bogged down in 'left brain' tasks such as setting up routing, or navigating the GUIs of different plug-ins to tweak one or two key parameters. As I said, I don't usually tend to use plug-in presets much, but go-to mixing chains — track presets, if you will — are different, and I could easily see myself creating my own StudioVerse library for key mixing tasks. I'd be able to work faster and, unlike with DAW track presets, could transfer them between the different DAWs I use. I was also happy to be reminded of some of the 'golden oldie' Waves plug-ins that I've fallen out of the habit of using. Best of all, though, I found the insights into how other (better!) engineers like to work really valuable.

The pros and cons of plug-in subscription packages will vary from person to person and I'll explore that another time, but StudioVerse certainly signposts one way manufacturers can add value to the more comprehensive offerings. Worth investigating.

If you wish to stick to using a preset's macro controls you can do so, but it's also possible to figure out what's going on behind the scenes.

summary

A vast online library of mixing signal chains that makes full use of Waves' extensive plug-in collections, StudioVerse offers a way to speed up your mixing and to learn more about the techniques of successful engineers.

£ StudioRack and StudioVerse are free, but require access to Waves plug-in collections: subscriptions start at $12.50 per month.
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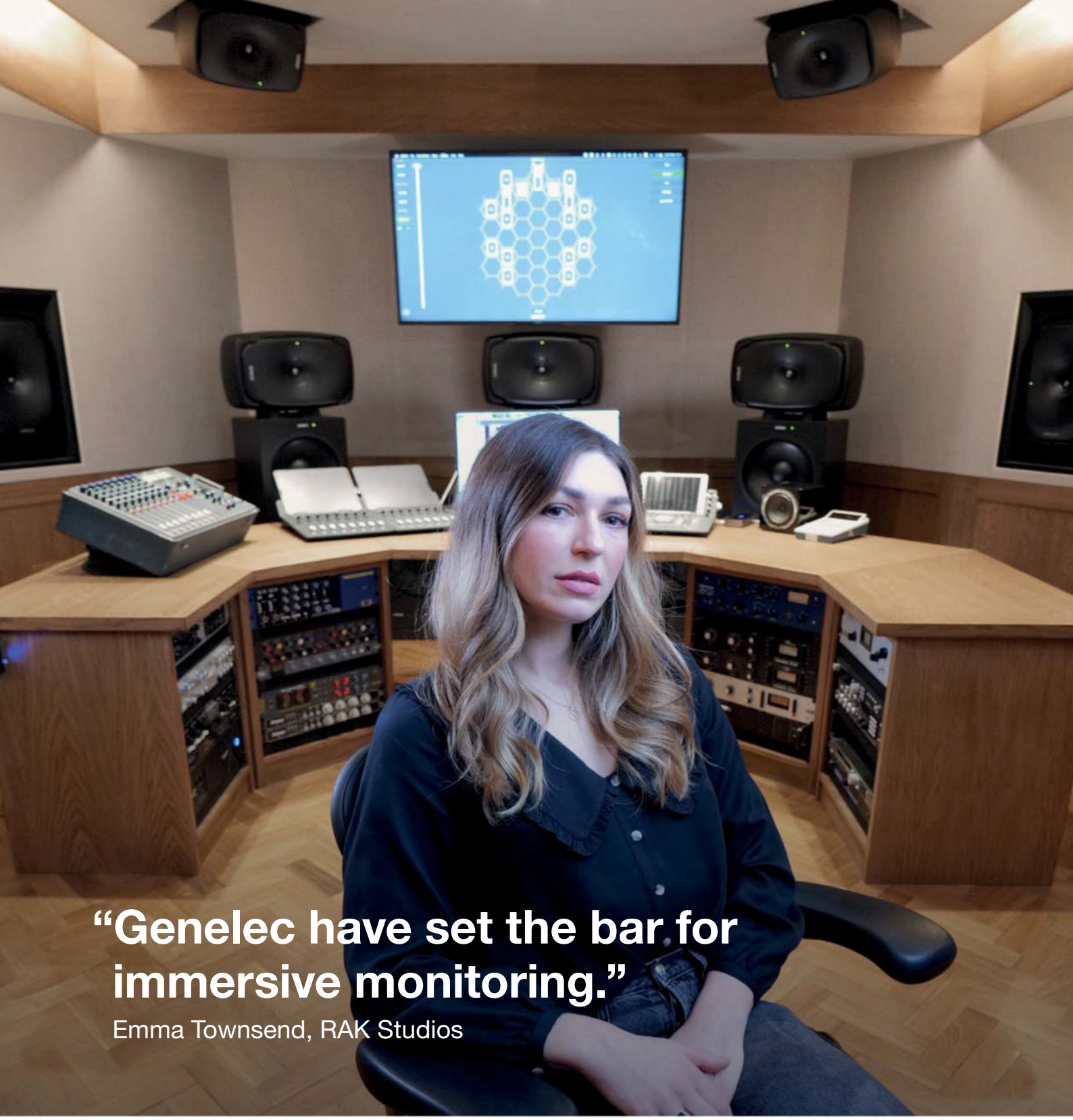

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THE CHANGE MAKERS

Meet the producers, engineers, songwriters and musicians featured on this month's *Sound On Sound* cover.

1. Anna Laverty
Anna Laverty is a producer, engineer, mixer and co-writer whose credits include Camp Cope, Stella Donnelly, Amanda Palmer, Pinch Points and Courtney Barnett. Anna is also founder of the Music Producer and Engineers' Guild of Australia.

2. Jennifer Decilveo
Jennifer Decilveo is a Grammy Award-winning producer, songwriter and multi-instrumentalist who has played a pivotal role in numerous projects across multiple genres.

3. Marta Salogni
Italian-born, Grammy-nominated recording engineer, producer and mixer Marta Salogni works from her own Studio Zona, in London. Some of the artists she's worked with include Björk, Depeche Mode, Sampha, MIA, Bon Iver, Black Midi, Animal Collective and Holly Herndon.

4. Manon Grandjean
Based in London, Manon Grandjean worked for award-winning songwriter and producer Fraser T Smith before becoming a freelance mix and mastering engineer in 2020. She has worked with a wide range of artists including Stormzy, Nao, Flo, Cat Burns, and many more, and has been involved in six UK number ones and 18 top 10s.

5. Halina Rice
Halina Rice is an electronic music producer and AV artist working at the intersection of music, art and technology: her sold-out live shows have been described as "part rave, part art-happening". Her performances and installations are frequently presented in spatial sound, both in physical venues as well as metaverse and VR environments.

6. Linda Perry
Beginning her musical journey as the lead singer of 4 Non Blondes, Linda Perry has profoundly impacted pop culture through her work as a producer and songwriter, helping shape the sounds of P!nk, Christina Aguilera, Alicia Keys, Britney Spears, Adele, Arianna Grande, Celine Dion, Miley Cyrus, James Blunt and many more.

7. Eve Horne
Award-winning creative mentor and advisor Eve Horne has over 20 years' experience in the music industry as a singer, songwriter and producer. She is founder of PeakMusicUK and the UNHEARD Campaign, and recently become an Ivors Academy Senator and a Board Director at the MPG.

8. Grace Banks
Grace Banks is a mixer and producer with a background in writing. She has engineered for Robyn, Jamie T, D Double E, Omar Apollo, Hozier, Jockstrap, Black Midi, Kelsey Lu and the Blessed Madonna, and has worked with producers and mixers such as Kid Harpoon, Inflo, Jim Abbiss, Alan Moulder, Catherine Marks, Markus Dravs and Marta Salogni.

9. Katie Tavini
Katie Tavini is a mastering engineer with a rich and diverse client list that is not limited to a particular sound or scene. Her recent credits include Bloc Party, Arlo Parks, Fatherson, Nadine Shah, Hundred Reasons, Emeli Sandé and Ash.

10. Emily Lazar
Grammy Award winner and eight-time nominee Emily Lazar has mastered more than 4000 albums for the world's leading artists, from the Beatles to David Bowie. Her studio, The Lodge, offers state-of-the-art mixing, mastering and specialised recording, as well as immersive mixing and mastering for Dolby Atmos. A proponent for gender equity and inclusivity, Lazar founded nonprofit organisation We Are Moving The Needle to help shape the future of the recording industry.

11. Catherine Marks
Producer and mix engineer Catherine Marks produced and mixed Wolf Alice's Grammy-nominated 'Moaning Lisa Smile', mixed the Grammy Award-winning album *Masseduction* by St Vincent and has production or mix credits on six Top 10 UK albums, including boygenius' recent number one album *the record*. Catherine is an Executive Director of the Music Producers Guild.

12. Ramera Abraham
Ramera Abraham is a London-based vocal producer, recording engineer and songwriter. She frequently works with music distribution and artist services company Platoon across their artist roster. Ramera was also included in SheSaidSo's Alternative Power 100 List 2021 as one to watch, and is the Music Producers Guild's Vocal Producer of the Year 2023.

13. Hannah V
Jumping from the Royal Academy of Music into playing stadiums with stars such as Rihanna, Jessie J, Jason Derulo and Taio Cruz, Hannah V soon found her true vocation in production and songwriting, working with artists such as Yola, Stormzy, JP Cooper, Lalah Hathaway and TOBi. In 2018 she scored her first gold-certified album from producing and co-writing on JP Cooper's *Raised Under Grey Skies* and in 2021 was part of the writing team for Yola's Grammy-nominated album *Stand For Myself*.

14. Olga FitzRoy
Olga FitzRoy is an award-winning recording engineer and mixer with credits including Coldplay, *The Crown* and the London 2012 Olympics. Recording Engineer of the Year in 2016, she served on the board of the Music Producers Guild from 2019-2022 and has previously stood for the UK parliament for the Labour party in Croydon, South London.

15. Leslie Gaston-Bird
Leslie Gaston-Bird began her term as President-Elect of the Audio Engineering Society in 2023. She is founder and director of Immersive and Inclusive Audio, CIC, which provides training in Avid Pro Tools and Dolby Atmos certification for underrepresented groups. Author of *Women In Audio* and *Math Fundamentals For Audio*, Leslie is Lecturer in Sound Recording & Music Production at City University in London.

16. Ann Mincieli
Ann Mincieli is a recording and mix engineer best known for her work with Alicia Keys. Mincieli has three Grammy Awards and five nominations. Ann and Alicia are co-founders of Jungle City Studios in New York, and partners in She Is the Music, a non-profit organisation aiming to increase the number of women in the music industry.

17. Steph Marziano
Philadelphia-born, London-based Steph Marziano moved to the UK to study at LIPA and later cut her teeth as an engineer recording Kasabian's UK number one 'For Crying Out Loud'. She now works as a producer, songwriter and engineer, and her co-writing credits include Hayley Williams (of Paramore), ODESZA and Bartees Strange.

18. Dani Bennett Spragg
Dani Bennett Spragg is a recording engineer and mixer who started her career as an assistant at Assault & Battery Studios, the home of Flood and Alan Moulder, in 2016. Dani is now freelance and works mainly from her mix room in West London. She was voted Recording Engineer of the Year at the 2021 MPG Awards.

19. Mika Sellens
Mika Sellens is a producer, writer, mixer and engineer from London with a background in electroacoustic music and sound art. Mika is a professor of Electronic and Produced Music at Guildhall School of Music and Drama, a Senator for the Ivors Academy, and an Executive Director of the Music Producers Guild.

20. Charlie Deakin-Davis
Charlie Deakin-Davies, aka charlieeeee, is a songwriter, producer and artist based at Ten87 studios in Tottenham Hale. Having been in-house writer/producer for Gary Barlow, Charlie has gone on to work with Laura Marling, dodie, MNEK, FRED, James Bay, RAYE, KAMILLE, GIRLI, Ellie Dixon, venbee, piri & tommy and more, and was twice voted NMG's Producer of the Year.

21. Fiona Cruickshank
Award-winning engineer, producer and film music editor Fiona Cruickshank has a portfolio that spans film, music and performance arts. In recent months, Fiona has worked with Alicia Keys and Keaton Henson, as well as with the National on their acclaimed album *First Two Pages Of Frankenstein*, while her film credits include *The Little Mermaid*, *Greatest Days* and *The Whale*.

22. Isabel Gracefield
Isabel is an engineer and mixer based at London's RAK Studios. She worked on Dua Lipa's multi-platinum, Grammy Award winning album *Future Nostalgia*, and other high-profile credits include PJ Harvey, Razorlight, Jim Abbiss, Lionel Richie and sessions for Hans Zimmer at British Grove.

23. Gili Portal & Stephanie De Angelis
Gili and Stephanie are assistant engineers at RAK Studios in London. Sound On Sound would like to thank RAK and Emma Townsend for making the studio available for our cover photo shoot.

This special issue of *Sound On Sound* is brought to you in association with the Music Producers Guild. Thanks also to Phoebe Fox and Andreia Lemos for photography and design.

Screen 1: The main view of Dolby Atmos Composer.

Fiedler Audio Dolby Atmos Composer

Immersive Mixing Software

Now you can create Atmos mixes using any DAW! But that's not all this app has to offer...

PHIL WARD

It probably won't have escaped your attention that mixing music for Dolby Atmos has become a big deal. Despite the inescapable fact that few music consumers out in the real world are fortunate enough to have listening spaces equipped with, say, 7.1.4 playback systems, there's a whole load of music being mixed for Atmos. There are several reasons for this. Firstly, the Atmos format is smart: an Atmos mix, which might have been created in a studio equipped with a full-fat 9.1.6 monitoring system, can, on playback, fold itself down to suit whatever channel format it encounters. Secondly, when an Atmos mix is created, the option is present to render a binaural version for headphone playback — you may have noticed that one or two folk out there primarily consume their music on headphones! Furthermore, Apple Music's Spatial binaural format is derived from Atmos encoded material, and both Tidal and Amazon offer Atmos binaural music streaming.

Back in the studio world, where Dolby Atmos mixes are created, the workflow has to date required an 'extended channel width' capable DAW and Dolby's not-inexpensive Dolby Atmos Renderer app. At the time of writing, the only DAW apps that officially support the extended channel widths that the Dolby Atmos Renderer expects are Pro Tools (Studio and Ultimate), Logic Pro, Reaper, Nuendo, Pyramix and DaVinci Resolve. (See the box for an explanation of what I mean by 'extended channel width'.)

Fiedler Audio's Dolby Atmos Composer completely upsets the trolly of fruit, because it enables DAW apps that only offer mono or stereo channel widths to join the Atmos party. But there's much more to Dolby Atmos Composer than that, because along with widening the range of DAW options it offers an Atmos mix workflow and routing philosophy that, to my mind, is better suited to the needs of music mix engineers than the somewhat prescriptive and convoluted demands of Dolby Atmos Renderer. Fiedler's Dolby Atmos Composer is also pretty affordable.

Screen 2: A stereo mix session created by the author for the review tests.

Dolby Atmos Composer is available from Fiedler Audio in two flavours; the fully featured version and a less expensive Essential version that omits a few of the more advanced features. This review is of the full version. The Fiedler Audio website has a full list of features not included in Essential, but I'll mention in this review if any feature isn't included in Essential. While I'm on the subject of software with two versions, closely associated with Dolby Atmos Composer is version 1.5 of Fiedler's SpaceLab, a multi-channel/immersive algorithmic reverb plug-in that's available in Interstellar and Ignition editions — this is primarily a review of Dolby Atmos Composer, but it's worth knowing that SpaceLab is fully integrated with it.

Objects & Beds

Before I get on to how Fiedler Dolby Atmos Composer can handle multi-channel widths in DAWs that can only do stereo, I need to introduce a further complication that Atmos brings to the multi-channel table: objects. Dolby Atmos is an 'object oriented' audio format in that any mono source can be described in terms of its position in 3D space defined by some associated XYZ coordinate metadata. In Atmos-speak, such an audio source is an 'object'. Objects can be placed anywhere in the 3D soundfield using a 3D panner plug-in and they can be dynamically moved around as a mix progresses with as much exuberance as you can imagine. You want that trumpet to circle around above the listener's head in time to the music? You'll want to make it an Atmos object.

It's substantially the concept of objects, with their positional metadata, that enables Atmos mixes to 'fold down' to match the abilities of the playback system. The Atmos decoder looks at the metadata and uses clever algorithms to work out how much of that trumpet, for example, should be sent to each available speaker in order for it to appear to be located as close as possible to where the XYZ coordinates say it should be. The more speakers in an Atmos replay (or monitoring) system, the more accurate the subjective result will be. However, Atmos doesn't simply employ objects, it also allows for audio 'beds' (Fiedler Audio use the term 'composite' rather than 'bed'). Beds are the Atmos equivalent of what we used to know as 'surround sound' — in other words, audio that's reproduced across multiple channels to surround the listener and create phantom images. There's no XYZ positional metadata encoded into beds, so their reproduced location is simply a result of traditional panning (although the panning can also include a height element).

An Atmos mix can contain a mix of beds and objects or be entirely created from one or the other, and the first head-scratching moments that I suspect occur for any mix engineer on entering the world of Dolby Atmos using Dolby's own Dolby Atmos Renderer app will concern session routing and how to accommodate beds and objects. It certainly was for me — I have quite possibly never scratched my head quite so much. Where once the paradigm was almost exclusively that a DAW output, bus or bounce is stereo, with the Dolby Atmos Renderer that is no longer the case, because not only are output busses much wider (the default Atmos bed for mix purposes is 7.1.2), there are also mono object tracks to accommodate that make their way, completely independently of the main output, to the Renderer. This all means that mix session routing and workflow using Dolby's Atmos Renderer has to be completely rethought. And it also means that any mix session that was originally configured for conventional stereo will probably have to be rebuilt from the ground up if it's to be mixed for Atmos.

A New Way

With the arrival of Fiedler Dolby Atmos Composer there's a new way of working, and it's one that's possible with a far »

Fiedler Audio
Dolby Atmos Composer
From $149

PROS
- Brings Atmos mixing to all DAWs.
- Enables simplified Atmos mix routing.
- Affordable.
- Intuitive and versatile in use.

CONS
- None.

SUMMARY
With Dolby Atmos Composer, Fiedler Audio have produced the first non-Dolby solution to rendering Atmos music mixes. But Dolby Atmos Composer also brings Atmos mixing to almost any DAW, and it offers routing options that make Dolby's own renderer seem somewhat Byzantine. It's Atmos mixing for the masses.

Screen 3: Dolby Atmos Beam.

» wider variety of host DAWs. There are two elements to Dolby Atmos Composer. The Dolby Atmos Composer plug-in itself and a second plug-in called Dolby Atmos Beam. Dolby Atmos Composer is inserted on the main output of the session, where it effectively sits in between the DAW and the nominated hardware interface. The output bus on which the DAC is inserted can be anything from a stereo channel width right up to 9.1.6. For the session I created for this review, I went with a stereo output bus to make it obvious from the screens that Dolby Atmos Composer doesn't need an extended-channel-width DAW! Speaking of which, Screen 1 and Screen 2 illustrate, respectively, the main pane of the Dolby Atmos Composer plug-in itself and the Dolby Atmos Composer mix session I created in Pro Tools for the review.

I'll describe Screen 2 first because doing so will explain the role of Dolby Atmos Beam, but first I'd like to give credit where it's due: the mix session incorporates stems from a track by a friend of mine, Danny Mulhern, and you can find the original stereo version of the track on Spotify at https://open.spotify.com/track/2RaFkY1RcasBLIW8BTsFvf?si=00fffa48b9a549bb. It's a piece that, to my ears, falls into an ambient contemporary classical genre that's well suited to an 'immersive' treatment.

You'll see in Screen 2 that I've bussed each audio stem to an aux track and inserted a send in each one for the Dolby Atmos Beam (DlbyAtmsB) plug-in. The role of Dolby Atmos Beam is to send the audio on the track directly to the Dolby Atmos Composer (DlbyAtmsC), bypassing completely the basic mix functions of the DAW. So the track's fader, mute, pan, solo and automation will no longer function. This is fundamentally how Dolby Atmos Composer is able to work with extended channel widths in DAWs that can't see beyond stereo — the actual 'mixing' is done in Dolby Atmos Composer rather than in the DAW.

The action of Dolby Atmos Beam to bypass the mix function of the DAW is why I chose to bus each stem to its own aux track rather than insert Dolby Atmos Beam directly on the stem tracks. Doing this means, firstly, that conventional fader moves and automation can be used to to tweak the mix, and secondly, it provides a neat way of creating a session that can be relatively easily configured for both traditional stereo and Atmos mix duties. In principle, you simply take an existing stereo mix session, add a second master output for Dolby Atmos Composer, and add an aux bus on which Dolby Atmos Beam is inserted as a send for each track.

You'll also see in Screen 2 that I set up a conventional reverb aux track (Stereo FX) that takes sends from all but the Horns and Vibes to create a reverb to be used in the Atmos bed. The reasons for leaving the Horns and Vibes out of the bed reverb send are, firstly, that I chose to assign those two stems as Atmos objects with their own reverb and, secondly, that I decided to use Fiedler SpaceLab reverb for them. SpaceLab routing works in a similar manner to Dolby Atmos Composer, with a Beam insert that sends audio directly to the reverb plug-in. Dolby Atmos Composer automatically identifies the presence of SpaceLab and includes it in its list of

Screen 4: The main window of the (separately available but deeply integrated) SpaceLab reverb.

Extended Channel Widths

By 'extended channel width', I mean a DAW in which individual track channels are not limited to mono or stereo, or even (as with Cubase) legacy surround formats such as 5.1, but one which can be configured to reflect immersive routing and output formats. Pro Tools Studio for example now offers channel-width options beyond stereo of 5.0.2, 5.1.2, 5.0.4, 5.1.4, 7.0.4, 7.1.4, 7.0.6 and 7.1.6 (Pro Tools Ultimate adds 9.0.4, 9.1.4, 9.0.6 and 9.1.6).

If you're not familiar with the nomenclature, the first digit defines the number of horizontal plane channels, the second digit defines the number of LFE channels, and the final digit defines the number of height channels. So, for example, the Dynaudio BM5 MkIII 5.1.4 monitoring in my studio comprises: left front, right front, centre, LFE (Dynaudio 18S subwoofer), left surround, right surround, front left height, front right height, rear left height and rear right height.

sources. And if you're wondering why the SpaceLab Beam tracks have an additional send (bus 17-18, set to 0dB) to the SpaceLab aux track, it's because, according to Fiedler Audio, a quirk of Pro Tools requires it for the correct management of plug-in latency. Other DAWS are unlikely to require the extra send.

And speaking of the Dolby Atmos Composer list of sources, Screen 1 illustrates the main pane of the plug-in. On the left is the list of Atmos Beam and SpaceLab Beam sources, each with a variety of options to configure; in the middle is a display of Dolby Atmos channels; and on the right are the Atmos output settings. Looking at the left-hand side of the pane first, the options include mute and solo (remember, they no longer work on the source tracks, although they can be automated by enabling plug-in automation) and a button that enables selection of the Atmos bed channel layout or enables sources to be assigned Atmos object status.

In the middle of the plug-in pane, the list of Atmos channels reveals that the first 10 are the bed channels, followed by up to 118 object channels. My mix session employs six object channels (11 to 16), the first two being the stereo Vibes, the second two the stereo Horns and the third two objects created by the SpaceLab reverb. I configured SpaceLab as stereo for the purposes of screenshot clarity but it can go all the way to a 32-channel full sphere. Each of these stereo sources requires two object tracks because in the Atmos world, objects are always mono. However, this raises some mix questions, because treating linked stereo channels as independent mono sources can potentially result in some unpredictable effects — it might actually make sense either to select the 'left channel only' mono option provided in Atmos Beam and SpaceLab Beam, or to create mono aux sends in the mix session. It seems to me a slight shame that Atmos Beam and SpaceLab Beam create mono sources simply by deleting the right channel.

Moving to the right-hand side of the plug-in pane, the configuration options cover output device and monitoring setup, and Atmos/binaural render export and loudness measurement. It's all very intuitive and easy to use, though having written that I should mention features that don't appear in Essential. This doesn't have the loudness measurement features and it's unable to monitor or export »

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» to headphones (binaural) and speakers simultaneously. One thing that it is important to emphasise, however, is that it's not only in terms of export that Dolby Atmos Composer can produce a binaural render of an Atmos mix. The monitor output can be binaural too. So it's perfectly feasible to create an Atmos mix without the luxury of, say, nine monitors and a subwoofer. It can be done with just headphones. Headphone-based Atmos mixing is of course also possible with the Dolby Atmos Renderer app, but Dolby Atmos Composer makes it decidedly more accessible.

Where The Magic Happens

Moving swiftly on to Screen 3, it shows the Dolby Atmos Beam pane and, in many respects, it's where the Atmos magic happens. The Beam plug-in not only sends audio to Dolby Atmos Composer, it also provides the 3D panning features that are perhaps the core creative element of all immersive audio. The panning display in 'full-fat' Dolby Atmos Beam can show a top, side or 3D illustration of the monitor positions and the listener, but in Essential only the 3D display is available.

The icons stacked vertically on the right provide options to constrain the panning axes. The audio sources are displayed in the panning display by the light-blue sphere icons — there are two of them because, in the case shown, the source track is stereo. The source icons can be selected by clicking singularly or in multiples, and once selected they can be moved around using the mouse. Alternatively, the azimuth, elevation and distance knobs can be adjusted. The spread knob increases or decreases the apparent size of an object in the immersive soundfield and the volume knob does what it says on the tin. In theory, the Atmos Beam volume knob could be used to replace the lost functionality of the source track fader. All the Beam panning controls, and the volume knob, can be automated by enabling plug-in automation and going back to the mix session to make the required adjustments in the automation lanes. Dolby Atmos Beam doesn't offer the panning tempo-sync options that are found in, for example, Dolby's 3D panner, but snapping automation break points to tempo markers can potentially achieve similar results.

Before I wrap up, while this isn't ostensibly a review of the SpaceLab reverb, users of Dolby Atmos Composer will no doubt be interested in it, so I thought it worth a quick discussion. Screen 4 shows the main SpaceLab pane, with a panning display that's not dissimilar to that provided in Atmos Beam. The orange blobs define the positions of the reverb outputs and, since I had SpaceLab configured in stereo mode for clarity's sake, there's just two. Screen 5 shows the SpaceLab reverb pane and, along with all the configuration controls you'd expect of a high-end reverb plug-in, there's a good selection of presets. The ones I tried displayed a natural and seductive smoothness combined with fine detail in the reverb tails, and the ability to spread such reverb fun around an immersive soundfield is genuinely inspiring.

Screen 5: SpaceLab offers all the controls you'd expect to find on a good algorithmic reverb.

Music To My Ears?

When I first began to experiment with mixing music for Dolby Atmos, about six months or so before conducting this review, I used Dolby's own Dolby Atmos Renderer. I'd naïvely imagined that I'd simply open an existing stereo mix and then just 'do' an Atmos version, but Dolby Atmos Renderer is primarily a commercial movie sound tool rather than a music mixing one, and it really was not that easy. In fact, I found the hoops it required me to jump through somewhat discouraging, as I'd spend more time working out and trying to understand the session workflow and routing options than I did on the creative possibilities of the mix.

Fiedler's Dolby Atmos Composer is significantly more friendly to work with in a music mix context, and had I started my immersive music adventures with this, I suspect my progress towards creating worthwhile (to my ears) Atmos music mixes would have been far quicker and rather less painful. This software doesn't yet do everything that Dolby Atmos Renderer can do. I've mentioned that there's no panning tempo sync, but there's also no monitor EQ or delay functionality so, the way I understand how these things work, an Atmos mix room based on Dolby Atmos Composer would be unable to achieve official Dolby certification. But that isn't necessarily a major problem, since it's possible to master mixes in a Dolby certified room, and I understand from Fiedler Audio that these features are in the pipeline. And for many prospective users, its compatibility with DAWs that can't handle extended channel widths will be a clincher. In fact, this could be a genuine breakthrough moment for Atmos music.

£ Dolby Atmos Composer €249 (discounted to €179 when going to press). Dolby Atmos Composer Essential €149 (discounted to €109). SpaceLab Interstellar €599 (discounted to €449). SpaceLab Ignition €249 (discounted to €179). Prices include VAT.
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LISTEN UP!

The Listener Profile: How We Perceive Music

Understanding the science behind musical taste and listening preferences can help you make more effective production decisions.

SUSAN ROGERS

I was in the studio nearly every day for 22 years. In the early days of my career, I was there to repair the equipment. Then, in 1983, I was called away from my tech job in Los Angeles to work for Prince in Minneapolis. He moved me from the tech chair into the engineering chair, where I stayed for several years. Returning home to LA, I engineered, mixed and produced records, mostly in the alternative/indie genre. Success followed thanks to the *Stunt* album with Canada's Barenaked Ladies. The money that music's royalty participants earned in the days before file sharing allowed me to jump across a divide and swim in a different stream: academia. For the next eight years I was in a classroom or laboratory, earning my bachelor's and doctoral degrees in auditory neuroscience.

Studying neuroscience turned out to be very similar to studying electronics. I saw parallels between the signal manipulations that we put our recording devices through and the way that our brains transform signals to become sound. Both kinds of audio systems have circuits, mechanisms, filters and processors for carrying out similar functions. While I was making records I focused on mastering how the studio worked, crafting sound through instrument choice, mic placement, equalising, compressing and monitoring. Learning to mix honed my engineering skills; learning to produce honed my mixing skills. Yet never once did I think about the mysteries of the most critical signal path of all — the distance from the ear to the brain. It is along this path that an acoustic

pressure wave becomes music. I'd have been well advised to consider it.

The exciting but daunting thing for record makers is that each listener is unique. A given record can reach two dozen ears as more or less the same signal, but in the fraction of a second that the resultant neural activity takes to reach the upper circuits of a dozen brains, it becomes something else. The record will be encoded, analysed, scrutinised, ignored, enjoyed, embraced, dismissed or remembered according to the listener profile: a collection of 'sweet spots' matching the features of music that elicit the greatest pleasure in the individual's brain.

Ears There & Everywhere

Listeners differ anatomically, starting at the pinna or external ear. I wish I had known in my record-making days how much the shape of our ears affects frequencies above 5kHz. It should have been obvious; simply push or pull on your ears in any direction and you'll hear a big difference in the frequency spectrum. Figure 1, a graph derived from the work of psychoacoustician Simon Carlile, illustrates the point.

The figure plots the head-related transfer function — the mediating influence of the ears and head on the sound spectrum — of seven subjects' right ears. Frequencies from 1-5 kHz get roughly the same 5dB boost in each case, reflecting the evolutionary advantage of detecting the most important sound in our world: speech. Above 5kHz the funny and complex shape of the pinnae makes individual differences more drastic. The participant whose ear is shown in purple has a big dip at 6kHz. The ear plotted in red has a typical 5-dB boost at 4kHz, which rapidly plummets to a 10dB cut at 8kHz. Comparing the light blue with the red trace at 8kHz, we see that the ears of these participants differ by 15dB. Imagine how these two listeners hear the high end in your mix!

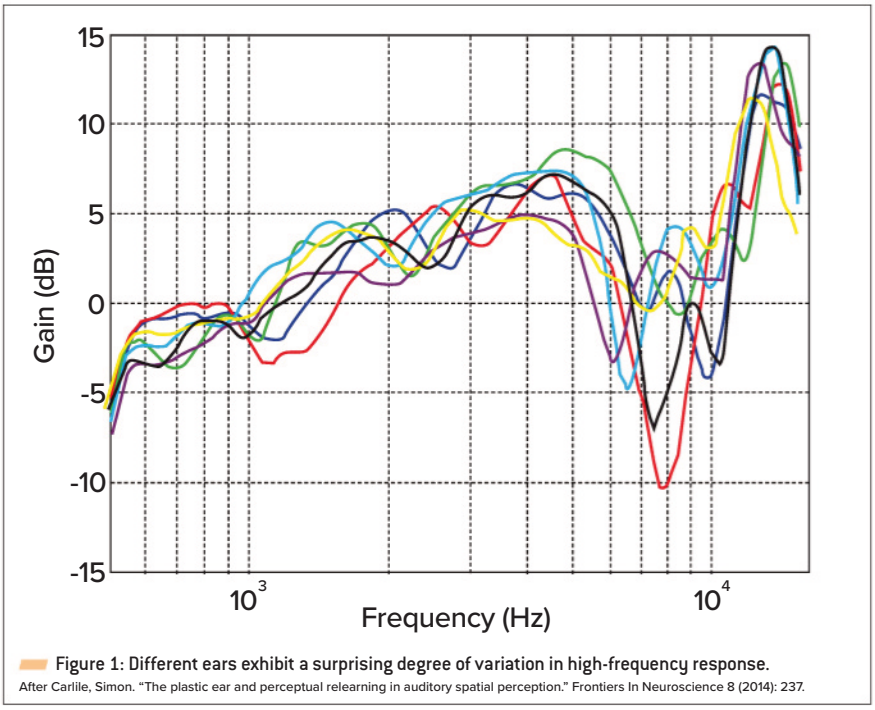


Figure 1: Different ears exhibit a surprising degree of variation in high-frequency response.
After Carlile, Simon. "The plastic ear and perceptual relearning in auditory spatial perception." Frontiers In Neuroscience 8 (2014): 237.

A Sense Of Belonging

Listeners differ motivationally, too. Our reasons for wanting to enjoy music vary throughout the day, and throughout our lives. Music's greatest asset is its functionality. We use music almost medicinally to match or change our mood, to focus, or to shut out unwanted thoughts. Our listening choices reflect our inner needs at any given moment. Over a longer time frame, our music library becomes 'self-congruous' in that its lyrics tend to reflect our personal beliefs and value system, or the culture in which we'd most like to belong. Try telling a young Taylor Swift fan that if she likes Swift's *Folklore* album then she'll love the music »

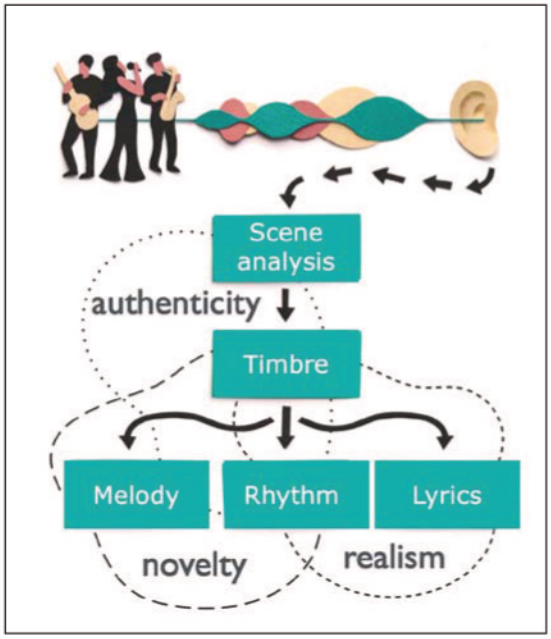
» of Maybelle Carter. I predict that the fan's reaction would be the same profound disinterest as that of my 15-year-old self when told that because I loved Led Zeppelin, I'd really love Howlin' Wolf. In my teens Led Zeppelin functioned for me as more than just music. The band represented my culture and sense of belonging. Now that I'm older and 'belong' with my new age group, I favour the Howlin' Wolf records.

Research in the overlapping disciplines of cognitive neuroscience, psychoacoustics and music perception shows what happens in our brains when we hear a record or a live performance for the first time. Immediately, our organ of hearing and auditory circuits pick out the musical elements from any environmental or speech sounds in the background. This is called auditory scene analysis. The ability develops over our lifetimes from exposure to different types of sounds. By adulthood, our brains can easily isolate the sonic signature of musical instruments and sung vocals. Once scene analysis has informed the brain that music is within earshot, timbre analysis lets us identify genre. Listening brains perform a snapshot survey of the different sonic make-up of electronica, classical and jazz music, for example. Neuroimaging studies have shown that many listeners' higher brain regions take less than a second to dismiss and ignore a record if timbre processing alerts them that it is in a disliked genre. If open to the timbre palette, we can focus on the musical and aesthetic features of what we're hearing.

Rhythm Is King

Past the gates of timbre analysis, internal 'wiring' delivers the sound signal to our auditory cortices, located above each ear. Each auditory cortex is a separate and equal processor. The left-hemisphere cortex is evolved to specialise in speech, while the right-hemisphere cortex deals with music. This lets us scan a record's melody and lyrics independently, hunting for the kind of musical treat that we personally enjoy. Lyric processing is faster than melody, but both of these are slower than rhythm: the fastest and easiest of music's features to comprehend. Record makers build short and long, strong and weak, syncopated and straight, behind-the-beat and ahead-of-the-beat accents into the rhythm track, suggesting how a listener might best synchronise her body with the music.

Maybelle Carter: the Taylor Swift of her age?



Once a listener has accepted the timbral palette of music, he or she will then start to appreciate its other qualities.

One of the things that sets humans apart from other species is how readily we can perceive rhythm. It may surprise you to learn that monkeys can't dance, at least not in the way that we do. Monkeys can't perceive rhythm because they lack our thick bidirectional neural tracts linking auditory and motor brain regions. These form a specialised loop running through the speech system. Listening to a groove that is 'just right' for your body sends an ancient social message to your muscles urging you to move along with it. This compulsion emerges when we are toddlers and is practically irresistible.

Past The Profile

If none of a record's musical features match a sweet spot on your listener profile, you can still find ways to like it. Aesthetic pleasure derives from a higher-order analysis of music's style and its relation to other pieces you have liked in the past. You can appraise a record cognitively and like it for its novel, innovative ideas or its traditional craft. You might appreciate the authenticity or sincerity you hear in its performances, or even in the performance restraint that only true masters of instrument and voice are capable of delivering.

Last but not least, there is the record's realism. For most people, record listening conjures up mental images: mind-wanderings that enhance musical pleasure. Being an engineer was a good career choice for me because my go-to visualisation has always been images of the performers. When I hear drums, I picture a drummer. When I hear singing, I 'see' the vocalist. I thought this was common until I conducted a research survey and learned that only around 15 percent of music lovers visualise the performers. Many listeners like to imagine themselves performing; others tend to favour abstract shapes and colours, and sometimes science-fiction worlds of imaginary planets. The most commonly described visualisation while record listening is autobiographical memories of people, places and events that were pleasurable in our past.

The kinds of images that we enjoy fantasising about influence the type of music that we like. My brain can find a cherished treat while listening to a realistic record played on familiar musical instruments, but it comes up empty if listening to electronica or techno, in part because I'm trying in vain to picture the sound sources. Many of my younger friends are just the opposite: they'd rather be carried away by programmed tracks and computer-generated sounds. Research suggests that fans of electronic music are less likely to experience autobiographical memories and more likely to see abstract or video-game-like imagery as they listen. New technology promoted today's musical trend toward less realism and greater abstraction. This is exactly what happened to 19th Century visual artists when the camera came along. Audiences for painting moved away from Rembrandt and towards Basquiat. Many people find joy in abstraction because it frees the brain from what has been called the "dominance »

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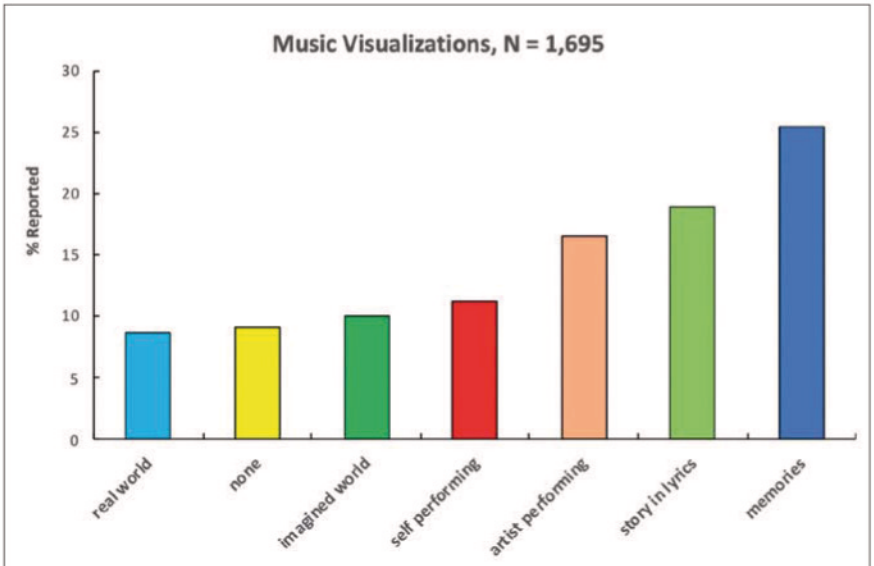


What do people visualise when they listen to music?
Rogers, Susan, and Ogi Ogas. *This Is What It Sounds Like: What The Music You Love Says About You.* W. W. Norton & Co., 2022.

» of reality", allowing the mind to construct its own interpretation of what the art is saying.

Disconnecting ourselves from the world around us and going into our own heads, so to speak, is something that we do a lot. According to research published in 2016, if you ask people if they are thinking about something other than what they are currently doing, they answer "yes" 30 to 50 percent of the time. When we disengage from the world to enjoy some spontaneous mind-wandering, we activate an interconnected set of nuclei labeled the 'default network'. The default network is the hub for our sense of self: self-consciousness, self-awareness and self-image. Because we are profoundly social creatures and music connects us to others, the default network is sensitive to music. A recent discovery revealed that when music listeners heard a record that they liked, their default network became more active. In contrast, when those same listeners heard a disliked record, a brain structure called the precuneus — a kind of gatekeeper for creativity and things concerned with the self — cut itself off from the default network, as if to prevent the disliked record from becoming associated with the self-image. Disliking certain styles of music feels like a knee-jerk reaction because, in a way, it is.

This is just one of many new findings in neuroscience providing support for the individualisation of the listener profile. Pleasurable listening experiences, whether enjoying music privately on headphones or at a concert venue surrounded by thousands of people, release dopamine. Even the adult brain is changed by experience, and feel-good neurotransmitters serve as the catalyst. The good feelings from music listening come from endorphins that actually hone the auditory system, refining it to become better and faster at recognising the 'music of you'. Your listener profile is different from everyone else's, and the older you are, the truer this becomes. The impossibility of satisfying every music listener should be intuited by record makers because it means that, rather than trying to hit numerous bullseyes with one dart, we can aim for our own musical centre: records that match our listener profile. Trust that nature and common experiences have created a wide variety of listeners, many of whom have profiles that at least partially intersect with your own.

The Profile & The Producer

How, then, can psychoacoustics inform record makers? Some of the things I learned in grad school made me wish that I'd had more insight during my time in the studio. Learning how the frequency spectrum is affected by the shape of our ears helped me realise how crucial it is to check our work on a variety of loudspeakers. Unless you are a highly experienced expert with years of mixing under your belt, you shouldn't mix exclusively on headphones. Headphone manufacturers must compensate for the absence of a head-related transfer function. If their filter design happens to be different to your body's own, your mixes will shape the spectrum to compensate, perhaps in the wrong direction.

Given how readily we respond to rhythm, it's important to follow Prince's advice: make your rhythm tracks "sound like a record". He believed that you should be able to mute all vocals and the melodic topline without losing any pressure. In other words, the rhythm track should be strong enough to hold the listener's attention. If your record falls apart when the vocal and lead instrument go away, keep working on your drum and bass sounds until they are exciting on their own. That way, if your melody and lyrics aren't as compelling as those of your competition, your record still has a fighting chance to earn some love. This applies to every type of music, even ambient, because our sense of time is an internal constant and we automatically seek to find a pulse or regularity in stimuli.

Know your audience demographic by actually watching them respond to your (or your client's) music during a live show. Note which songs are most likely to get them moving, and make special note of how they move: up and down, side to side, and even front to back. Build that motion into your rhythm tracks. Men and women favour different motions due to the way that our weight is distributed. Note which pieces in your set draw which gender closer to the stage. Consider a song's lyrical message in tandem with its rhythmic message if you want to speak to an ideal listener.

Accept that your listener's brain has plenty to keep it distracted and amused. If you want its attention, assume that you need to work harder for it. Mix until you think you are done and then enhance the exciting elements 10 percent more. Your work will have to overcome a brain's many competing signals and internal drives, so be even more arousing than you think is warranted for a given piece.

On the surface, an abstract painting and a four-year-old's scribbled drawing might look the same, but research shows that viewers can accurately identify which is which. The painter's gestures are the outcome of his intention; even non-experts pick up on the subtle cues that caused lines and colours to appear just so. As viewers can see intentionality, so listeners can hear it. When you are in the studio, listen for the underlying meaning in every gesture. If your instinct tells you that the singer or player isn't genuinely feeling the performance, don't think that no-one else will know. Have your performers put some weight or purpose into every note.

You don't have to be an art expert to instantly recognise the differences between abstract art done by trained adults, and a child's painting!

Knowing something about the neuroscience of listening will not (and should not) change the in-the-moment running stream of decision-making that you employ while making a record. Creative work demands a heavy contribution from instinct, a sense of knowing what will work. Analytical work — understanding why something worked or didn't — should happen offline. But it should happen. Make that offline analysis easier and even more instinctual by getting better acquainted with your personal listener profile. Savour records deeply so that your auditory cortex can morph into a perfect receptacle for the 'music of you'. Know the timbres, rhythms, lyrical style, melodies, harmonies, genres, sound designs and performance gestures that resonate the most powerfully for you. The best work you'll do will be in service of your listening brain as it hunts for or creates the musical treats you crave. Trust that although no two listeners experience a record in the exact same way, your work can offer a menu of features rich enough to deliver a rewarding musical meal to a broad range of consumers.

Modartt Pianoteq 8

Modelled Virtual Piano

Modartt's modelled piano goes from strength to strength and is now, remarkably, available for iOS.

ROBIN BIGWOOD

Pianoteq 8 has been out for over half a year now, so it was high time for a careful look at it in *SOS*. The more recent news, though, is the release of Pianoteq for iOS and iPadOS, supplementing the existing Windows, macOS and Linux versions. It's good to see really mature, sophisticated apps being released for the platform, and we'll assess it here in parallel with its desktop counterpart.

In case you're new to Pianoteq, the main need-to-know feature of this virtual instrument app and plug-in is that it offers the sounds of a wide range of acoustic and electric pianos, harpsichords, mallet instruments, pans, harps and more, but unlike most of the competition doesn't use samples. Instead it works by modelling, computationally, the mechanics and acoustics of its included instruments in real time. Consequently the installation size on your computer is staggeringly small, at just 50MB. In the early days of Pianoteq, more than a decade ago, the corresponding 'gotcha' was whether your CPU could handle the maths; nowadays any half-decent recent chip easily will. So Pianoteq's modest demands on your computer are out of all proportion with the instrument library depth and notably fine playing responsiveness it offers. The computational basis also opens up possibilities for creative sound design that leave many sample-based pianos far behind. Certain other aspects, like sympathetic resonance and subtlety of pedal behaviour (including a dedicated Celeste pedal, that can turn any piano into a felt-hammered equivalent), are unparalleled.

Pianoteq 8's new features are not Earth-shattering, but they are diverse, and valuable. There have been general improvements to the sound and behaviour of all modern-era acoustic pianos, the reed electric pianos and the Concert Harp. The user interface has been freshened up, with a new pictorial 'shutter' over the slightly esoteric Tuning, Voicing and Design parameter panels. Finally there's a new €49 add-on instrument pack, a classical guitar, which for the first time features playback features similar to sophisticated scripted sample-based instruments. And then there's the iOS/iPadOS version (which I'll just refer to as iOS from now on).

In i Places

I did equal amounts of testing with the iOS and macOS versions of Pianoteq 8 (in its Pro tier guise, which offers the most in-depth sound editing). On desktop installations v8 happily exists alongside early versions, so any existing DAW projects will be unaffected. Downloading and authorising the app on iPad was quick and straightforward, via the normal Apple App Store channels: the 'Restore Purchases' feature is used to tie in to any existing Pianoteq 8 licence on Modartt's own servers, and an iOS unlock will use one of the three authorisation seats provided as standard. Although interestingly, using it on both my iPhone and iPad still appeared to only use one seat, so the authorisation count would seem to be on a per-account rather than per-device basis.

Pianoteq on mobile is, then, almost identical to its desktop counterparts. Certainly in all the ways that really matter: sound quality, responsiveness, availability of all instrument packs, provision of parameters for sound design, and even the Standard and Pro tiers' Note Edit and Morph/Layers features. The only thing that really differs is some aspects of the user interface. On desktop the Pianoteq window is still close to square in shape; on oblong iPad and iPhone screens

Modartt Pianoteq 8
From £119

PROS
- Noticeable improvements to the sound of all modern acoustic pianos, plus Wurlitzer and Concert Harp.
- Flawless, full-fat implementation as an app and AUv3 plug-in for iPhone and iPad.
- New guitar instrument add-on sounds phenomenal and is easy to play from a keyboard controller.
- Cements Pianoteq's position as one of the best virtual piano options out there, with staggering sound design possibilities.

CONS
- Quality never comes cheap: iOS musicians might need to brace themselves for this app purchase, unless they can benefit from desktop version upgrade pricing; collecting additional instrument packs gets expensive too.

SUMMARY
A valuable update to an already sophisticated virtual instrument, Pianoteq 8 sounds and looks better than ever before, and is available to iOS users for the first time. It has also taken up the guitar...

Guitarteq

Pianoteq has for a long time been more than just a piano library, and with the new Classical Guitar instrument pack it becomes more flexible again. Hollow, semi-hollow and solid-bodied timbres are available.

It's possible to play Pianoteq's guitar as if it was a piano, but that would be missing the point a bit. Instead, a new guitar playback mode interprets your playing and intelligently apportions notes to the six virtual strings, which can be set to a standard tuning, drop D, or various open alternatives. You can have Pianoteq make all voicing decisions, or skew left-hand position higher up or lower down the fingerboard — with corresponding changes in tone colour — using MIDI keyswitches. The six strings can also be made to respond to separate MIDI channels.

Additional keyswitches trigger harmonics (flageolet), a rasgueado mode, a thumb pluck, glissandos (via legato note overlaps), slide-ins and outs, palm mutes, restrikes of held notes and left-hand position shifts. A slider controls how likely you are to hear string squeaks and other non-note sounds.

The sense of realism here can be staggering, even if you know nothing of guitar fingering and only stick to the main timbre, which is jolly convincing. Go deep into the keyswitches and it gets even more so. A few aspects of implementation are curious, like the Easy voicing mode sometimes not allowing intervals to sound that are clearly easy to finger, on adjacent strings, without stretching. But on the whole this is an amazing new development and a serious alternative to the many sample-based alternatives out there. It sounds as good soloing as it does playing arpeggiated figures or chords. The basic nylon-string timbre can take the onboard Amp effect well to tease out some jazz guitar and lightly overdriven timbres. Presumably there's no reason we mightn't see steel-strung acoustics, Strats, Teles and Les Pauls in the future.

the so-called 'instrument' and 'audio engineering' sections scroll up and down (with prompts to show you how when you first start the app). Additional views like a large keyboard and the MIDI file player are exposed by various swipes and drags. Equally though, if you open the Pianoteq 8 AUv3 plug-in in a host like Kymatica's AUM, it does appear square there, just like the desktop version.

One other slight difference on iOS is the scheme for storing user presets. On iOS, presets are saved deep in the file system, out of reach of the Files app. You can liberate them via a Share command from the dedicated preset browser, and load new ones as FXP files.

Because Pianoteq is not replaying samples it offers unparalleled flexibility for tweaks and programming of various kinds. Shown here is its multi-channel virtual miking feature and, in the next screen, the Note Edit view, letting you adjust parameters on a per-note basis.

The performance of the iOS Pianoteq is remarkably good: it ought to be, of course, running on Apple A- and M-series chips that are as powerful as many good laptop and desktop computers. I still occasionally had to pinch myself though, that here was Pianoteq, not watered-down in any way, running on my phone... Modartt recommend "a recent iPhone/iPad model" but say it runs well on older models using iOS 15 or higher, including the venerable iPhone 6s, iPad Mini 4 and iPad 5.

On my fifth-generation iPad Air, with an M1 chip, I never saw more than about 14 percent processor usage reported by Pianoteq itself, even when deliberately sustaining many low piano notes, which is computationally intensive. That was with a 128-sample audio buffer, and 2.7ms latency. On an iPhone 12 from 2020, with its A14 chip and the same settings, it wasn't much worse: 25 percent at most.

All things considered, Pianoteq is gold standard iOS software. It's easy, quick and intuitive to use, works beautifully via a touch interface, and has massive musical potential and programming depth. It's a lot more expensive than most apps, but, well, you get what you pay for.

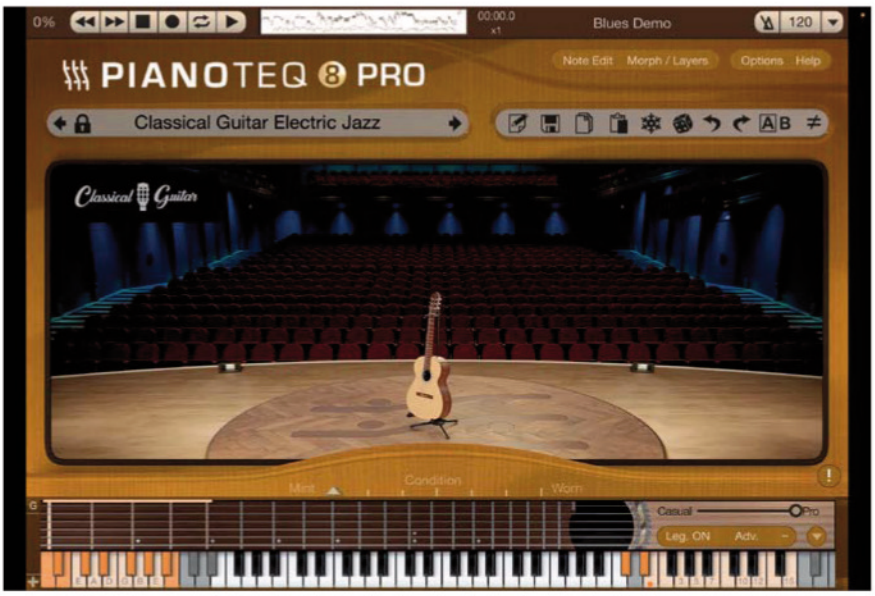
Levelling Up

To get a sense of Pianoteq 8's improved piano voicings, I went back to the Mac, where I could directly compare v7 and v8. I made presets in Pianoteq 7 and opened them alongside in Pianoteq 8. Then, after repeatedly rendering a single MIDI region, for a cross-section of pianos and »

How To Buy

Pianoteq 8 comes at a range of prices, according to what you want from it. The most affordable option is the Stage version, at £119, which lets you choose two instrument packs but limits scope for creative tweaking. That price is the same for desktop and for iOS, via an in-app purchase. The Standard version (£219) gives you three instrument packs, and opens up the vast majority of possibilities for sound tweaking, tuning, virtual mic placement and Morphing/Layering. On iOS you upgrade to Standard in-app, for the same total cost. The Pro tier at £319 lets you adjust almost all main parameters on a per-note basis, offers full overtone editing, supports sample rates up to 192KHz, and comes with four instrument packs.

Any Pianoteq version can extend its sound library further through additional instrument packs: all are £45. If you're a have-it-all sort of person though, you might want to explore the Studio bundle from the outset. It's a considerable £699 but gives you everything: Pianoteq Pro plus all currently available instrument packs. All prices include VAT.



Pianoteq on iPadOS, with the new Classical Guitar instrument pack loaded. The keyboard graphic area shows labelled keyswitches and indicates how played pitches are being fingered.

» the other upgraded instruments, I could make a more meaningful and seamless A/B comparison.

These things are subjective of course, but Pianoteq 8 sounds significantly better to me. The various Steinways plus the excellent Grotrian I tested in depth all benefit from a smoother overall sound, less middle-dominated and insistent, and with somehow more finesse and complexity overall. At higher velocities strong attacks seem to integrate better with the later decay phase too: this was not bad before, but it's more coherent now.

The difference in Pianoteq 8's Wurlitzer, 'Vintage Reeds W1', is much more pronounced. It's got loads more 'girth' than in v7, is actually just more bassy, and like the acoustic grands is less liable to a congested tone in the upper mids. I don't always go to virtual Wurlitzers expecting much sophistication or timbral depth, but this has it. It's quite something.

Then, by contrast the Concert Harp is now less bass-tilted. It sounds more precise and transparent now, in a good way, with a more extended top end and detailed attack phase, which very nicely communicates the sound of fingers gripping and leaving strings.

There are one or two interesting measurable differences between Pianoteq 7 and 8 too. Bouncing MIDI regions for the acoustic pianos in the latest version of PreSonus Studio One, Pianoteq 7 would often render at 12 times faster than real time, and not change much through the process. For Pianoteq 8 I'd see a similar figure for a moment, but then the render would drop to nearer four times real time. That implies that more maths is being done, which is in line with

the claim of increased sophistication in the physical model. The harp, incidentally, went through equally fast, at 20 times faster than real time on both versions, so it's a mixed picture.

Great 8?

Like many readers, I'm sure, I've been fascinated by the development of Pianoteq over the last 17 years: on a pure technological level, but also for what acoustic modelling can mean for us in terms of expressivity, usability and flexibility.

Version 7 was, I thought, a big moment for Pianoteq, delivering truly classy-sounding pianos alongside fascinating new possibilities for sound design via Morphing and Layering features. My review from April 2021 (www.soundonsound.com/reviews/modartt-pianoteq-7) goes into depth about that. Version 8, simply, is a good step better again.

The fact that Pianoteq can now be on your iPhone or iPad allows for even greater flexibility in how and where it can be used: that tiny installation size now makes more sense than ever. We've grown used to paying peanuts for apps, and in that context the €139 in-app purchase cost to enable new purchasers to unlock the free demo might seem unreasonably high. Bear in mind though that there is almost nothing like it in the App Store. Yes, there are more virtual pianos there than you can shake a tuning fork at, but almost all are toys in comparison. The few exceptions include Synthogy's Steinway soundsets for Korg Module Pro, and UVI's Ravenscroft 275, but these are single pianos with limited scope for creative treatment, approx 1GB installation sizes, and nothing like the sophistication of Pianoteq. Also if you own any previous desktop version of Pianoteq then the cost of v8 is actually €29: that's the single, across-the-board upgrade fee to get up to date, and at that point you can unlock the mobile app for nothing. If you're able to benefit from that it's a heck of a bargain.

Alongside Pianoteq 8's flexibility, there's a new standard of quality: it's both rewarding to play and nicer than ever to listen to. Acoustic grands have improved significantly. The Wurlitzer and Concert Harp have both leapt forward in complexity and authenticity. There is very little not to like. Complaints about sound quality are now more a matter of taste than anything, and because you can try out all instrument packs in demo mode at any time there's no reason to end up with anything you don't want.

Then the Classical Guitar instrument pack. Not everyone will need or want this, but if you do then it's a state-of-the-art virtual guitar that sounds uncannily realistic.

The conclusion couldn't be simpler. If you're eligible to upgrade to Pianoteq 8, just do it. This is fabulous software that has reached new heights of sophistication. If you're investing for the first time, it's a great time to get on board.

£ See 'How To Buy' box.
W www.modartt.com

Kurzweil SP7 Grand

Stage Piano

Kurzweil's new stage piano makes the case for simplicity as a virtue.

GORDON REID

There was a time when almost all electronic stage pianos were horrid affairs capable of generating a small range of sounds that sounded nothing like their namesakes. Then Kurzweil changed the landscape forever with the sample-based piano in their K250. Soon after, the market was awash with digital pianos from the likes of Roland, Kawai, Korg and Yamaha. A decade or so later, workstations got in on the act and, while early attempts lacked the quality of dedicated instruments, this changed in the 21st Century, and the choice of keyboards offering high-quality acoustic and electro-acoustic piano emulations is now extensive.

Throughout all of this, Kurzweil held their heads high. Following on the heels of the K250 plus its K1x00 spin-offs, VAST synthesis appeared in 1991 and, for more than 30 years, almost all of Kurzweil's synths and pianos were based upon this. Indeed, recent Kurzweil pianos such as the Artis (*SOS* May 2014) and Forte (*SOS* November 2015) offered access to the full VAST synthesis engine including its KB3 organ extension. But last year, the company announced a pair of 88-key stage pianos — the SP7 and SP7 Grand — based upon a hugely simplified sound generator that they call ATST (Authentic Timbre Synthesis Technology).

You probably want to know what the differences between the two models are and, to my surprise, there are not many; the Grand sitting beside me has a metal case that makes it feel more robust than the plastic SP7, it uses a Fatar rather than a Medeli keybed, it boasts a 7-inch touch-sensitive screen rather that the cheaper model's 4.3-inch screen, and it's black not white. In the other direction, the SP7 has four cursor buttons and an Enter key deemed superfluous on the Grand. Then there's the... umm, well... no, there's not. As far as I can see, those are the only differences between the two models, so the rest of this review is applicable to both.

Introducing ATST

ATST itself draws upon Kurzweil's huge sample library but, instead of allowing you to edit and control everything you can think of (and some things that you probably haven't) it offers a selection of preset sounds that you can tweak to taste. It has also dispensed with the common Patch/Multi architecture and replaced it with a single programming environment called Sound Mode, which is, in essence,

Kurzweil SP7 Grand
£2280

PROS
- Its simplicity may be attractive to players who don't want to delve deeply into sound design.
- Editing on the Grand's 7-inch screen is fast and simple.
- Like all Kurzweils, it can sound very nice...

CONS
- ... but it can suffer from audible velocity switching and sample inconsistencies across the keyboard.
- Important MIDI editing facilities are conspicuous by their absence.
- You can't use the performance controls to modify any parameters within the effect units.
- You can't control the key or scale of the vocal harmonising.
- No aftertouch.
- There are operational glitches that suggest that an OS update is needed.
- It uses an external power supply.

SUMMARY
Despite being a product of one of the most loved and highly respected brands in the keyboard world, I have a feeling that the SP7 Grand may have missed the mark. It will be interesting to see how it develops, as I feel it must.

a 16-part multi-timbral... umm, Multi. A number of manufacturers have dabbled with this approach in recent years and I wish that they would stop — it doesn't make it easier to understand or program an instrument, and I would venture that it makes it less so.

You can place up to 16 presets in a given Sound. Each occupies a Zone with its own key and velocity ranges so that you can create complex multi-timbral setups as well as lush layers. The presets are drawn from 14 categories: pianos, e-pianos, organs, brass and so on, plus a User category into which you can save your own tweaks. Unfortunately, there's no way to create categories into which you can, for example, save Sounds for specific projects.

Much of the Zone editing is aimed at performance; for example, you can select the destination and value range of each physical controller for each of the Zones. The choice of destinations looks impressive but, as I later discovered, it's far less flexible than I had assumed and it lacks many capabilities that I had expected. You can also modify the presets themselves, although the range of parameters is tiny: velocity response, a simple ADR contour generator, a filter with controls for cutoff frequency and a mild resonance, vibrato, portamento, fine-tuning and pitch-bend. You mix the Zones using the Channel Mixer, and this provides access to the pan and volume for each preset as well as its sends for the four insert effect busses and the main chorus/delay and reverb effects. Unfortunately, the manual offers neither a full description nor a block diagram of the effects architecture, so I fear that (like me) you'll have to experiment to find out what's happening.

Each insert effect bus offers four pre-defined slots — dynamics, EQ, modulation/wah, and delay/tremolo — that can host your choice from 21 editable effects algorithms. You can only pass the sound from a given Zone down one bus at a time so you can't use the modulations slots in two busses to apply (say) phasing and flanging to the same sound. But, despite the limitations, you can obtain some pleasing results. There's also a four-slot auxiliary effects section that, in my view, should be named the Master Effects section. You can determine whether or not each Zone and the output from each effect bus is passed through this, at what level, and how the delay, chorus, reverb and EQ within it interact.

To spice things up a bit, there are eight latchable arpeggiators that can act in the traditional fashion or as simple step sequencers. If the former, three modes, three tempo ratios, four swing settings, four gate times and two velocity modes are provided, and you can also determine the range of notes within which the arpeggio will be played. Alternatively, you can use the step-sequencer grid to program patterns of up to 32 steps. In addition to the type of event occurring on each step, you can choose which note in a held chord is affected, and define its pitch and velocity. Although it's quite a simple system, some impressive results — including rhythms and accompaniments — can be obtained.

In Use

It didn't take me long to get to grips with the Grand once I had worked out its effects structure. The touch-sensitive screen was a large part of the quick learning curve; in general, its menus are well designed and, although some of the icons are grouped rather closely, selecting the wrong one became relatively rare once I had discovered that you have to touch them slightly above where they appear on the screen. »

The Rear Panel

As befits a stage piano, the rear panel is not particularly dense. It begins with dual audio inputs that accept both XLR and quarter-inch plugs and feed the audio input mixer, input effects and vocal processor. Next to these you'll find quarter-inch main and quarter-inch monitor output pairs, and a quarter-inch headphone output. There are three pedal control inputs: two switches that default to sustain and sostenuto, and also offer half-pedalling if you buy a suitable pedal, plus a volume pedal. An SD card slot is provided for updating the firmware, and a USB port (which can also be used for updates) carries both MIDI and bidirectional stereo, 24-bit/48kHz audio. Audio received from a Mac or PC can be passed through the effects and returned to the computer in real time. Traditional 5-pin MIDI in and out sockets (but no thru) are provided, and the final hole is for the output from the external DC power supply. Inevitably, I'm not happy with the use of an external PSU; having one hang off the back of a £2000 instrument designed for stage use is a surprising choice, especially since there's no cable restraint for the barrel plug. What's more, the PSU delivers 2.5A at 15V, so it's unlikely to be something that you'll be able to replace at the local electrical store if you forget yours at home on the afternoon of a gig.

In contrast, I must admit that I didn't much like its Fatar TP/110 keybed, although switching the action from the factory default to a lighter one and tweaking the velocity responses of some of the sounds was a significant improvement for many of the piano and e-piano presets. But I never fully made friends with it. Let me explain... In addition to all of its other non-piano sounds, the Grand includes a selection of drum kits, but I found it hard to play these rapidly without missing beats. Of course, a piano keyboard isn't ideal for organ swipes and synth soloing either, reinforcing the view that Kurzweil see the Grand as a digital piano with bonus sounds rather than an as all-encompassing keyboard.

A small, programmable joystick takes care of pitch-bend and modulation duties. I have no gripe about this choice of controller because omitting a performance panel to the left of the keybed helps to keep the instrument manageable, but it's not ideal for some performance purposes. Another thing that bothered me slightly was the choice of three toggle switches to control the audio inputs (see box). These look and feel incongruous and I can't say that I'm a fan. But these are minor quibbles compared with the omission of aftertouch, which is becoming far too common on expensive instruments. Sure, it would be an irrelevance on a dedicated piano, but the Grand offers all manner of additional sounds that would benefit from it.

Polyphony is a generous 256 voices although, unlike previous Kurzweil pianos that included the KB3 engine, its sample-based organ sounds eat into this when you use them. And, in common with all keyboards that offer Multi modes, layering presets reduces the polyphony proportionally.

Given the reduced range of sounds on offer, the Grand's ROM is significantly smaller than that of other modern Kurzweils, although I was surprised by the scale of the reduction — just 2GB here versus the 4.5GB of the K2700 workstation and the 16GB of the Forte. Clearly, its library comprises a lot fewer or shorter samples (or both), especially since Kurzweil claim higher resolution in the Grand. Having said that, the specification suggests that the higher resolution is used in the maths of the filter rather than in the storage of the underlying PCMs, so who knows what's going on?

There are 301 factory programs and 512 Multis (the company's terminology is not wholly consistent) with 128 User memories and 128 slots for Favourites. This seems rather parsimonious given the cost of memory in 2023, so it's just as well that there's an editor/librarian in preparation. I would have liked to have included a few screen grabs of this but it was still unavailable at the time of writing.

I started my audio tests with the grand pianos and, if I'm honest, I wasn't a fan of these when I first played them because they sounded rather flat and uninspiring by modern standards. So I set about hunting for a reason and soon found it; both the string resonance and damper resonance are defeated in the factory settings. Switching these on and adjusting them to suitable values added depth, warmth and playability, and I was much happier. Had I invested in a half-damper pedal, I would no doubt have been happier still. Nonetheless, as on the K2700, which supplied the flagship pianos for the Grand, they weren't entirely to my taste. They can sound great when playing rapid, percussive passages, but for slower — dare I say it, romantic — compositions they lack the deep, rolling bottom end of my favourite grand pianos and their best emulations.

I then tested the electric pianos. The underlying tones of many of these were nice, but the velocity transitions could be quite brutal in some of the key ranges of some of the presets. So I moved onto the Clavis and harpsichords and encountered uneven sampling (meaning that the tone of one key could be somewhat different from that of an adjacent one) in addition to velocity switching. Many of the strings also exhibited audible velocity transitions, as did... well, I don't need to belabour the point further. Of course, velocity switching and uneven mapping are far from unique to the Grand, and you'll encounter them on almost every PCM-based instrument. But whereas some modern synths and workstations allow you to ameliorate any adverse effects using facilities such as velocity crossfading, ATST does not.

I moved on to the organs to see whether a small selection of sample-based sounds

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» could replace the KB3 organs of other Kurzweils. I started playing — hmm... not too bad, I thought — and was comfortable with the lack of any means to alter the registration because that's the nature of the beast. But then I tried to change the speed of the rotary speaker effect. Then I tried some more. It didn't occur to me that this wasn't possible so I checked every page in the menus and every page in the manual but eventually had to concede that it's not possible. You can program the effect to be fast or slow in any given preset, but you can't switch between speeds. This renders the rock/pop organs almost useless for me. Unfortunately, there's nothing in Kurzweil's blurb to suggest that KB3 (or a modern equivalent) will be reinstated in the Grand, nor is there the top-panel hardware to support it.

Having discovered the limitation with the Leslie emulation, I checked whether any of the other effects allow real-time control. Unfortunately, none of the parameters within the insert effects can be assigned as controller destinations. You can only affect their output volumes, pans, and send levels to the master chorus/delay and reverb. I was surprised.

Unfortunately, I also experienced some operational glitches while conducting these tests. In no particular order, these included a locked screen and locked editing controls when modifying the effects on a 12-string guitar preset (the only cure for which was a power cycle), a sitar preset that suddenly developed a noise like a helicopter behind it (a factory reset was needed to eliminate this) and a short but deafening burst of digital hash that I thought was going to blow my tweeters when I was editing the aux effects. In addition I noticed that, when you select a new Sound with the damper depressed, the Grand maintains the existing Sound until you release the pedal (which is good) but that, if you're still playing when you release the pedal, there's a glitch. (I'm guessing this is caused by the effects processors reinitialising.) A quick check showed that the review unit was running the current OS, so I couldn't blame any of this on pre-release software. Nonetheless, I started to wonder if the Grand had been rushed out before all of the wrinkles had been ironed out. Whether it has or not, I hope that these prove to be transient problems that will be eliminated in the update that Kurzweil assure us is forthcoming.

Something else that needs addressing is the lack of MIDI editing. You can determine whether each Zone transmits to the local sound generator or over MIDI or both, but that's all. You can't even change the MIDI channel number: Zone 1 is always MIDI channel 1, Zone 2 is MIDI channel 2, and so on. What's more, if a Zone is transmitting to the outside world, it always sends Program Change messages when you scroll between Sounds or insert a new preset, which is infuriating. Similarly,

Vocal Processing

The Grand offers two audio input channels that allow you to mix and process external sources, either for keyboard karaoke or to add vocals to your performances. You could even use them to turn the keyboard into a three-channel mixer for small performances.

Each channel offers a useful range of input effects including gating, dynamics, rumble filtering, EQ, modulation, delay and tremolo. The output from each can also be passed through the instrument's chorus/delay and reverb processors. But perhaps the most powerful facilities that you can apply to any external audio are pitch correction and four-part harmony generation. There's a lot of flexibility here, with formant shifting (emphasising male or female vocal qualities), pitch transition smoothing, delayed vibrato, and all manner of harmonisation including major and various minor keys, 10 voicing variations, humanisation (slight pitch inaccuracies in the generated voices) and portamento. The output quality is also rather good, so there's a lot to like.

However — and this is a huge however — there doesn't appear to be any way to make the harmonisation follow what you're playing so, for example, harmonies in C melodic minor at the start of a song remain in C melodic minor no matter how it develops. This renders the harmony generation almost useless. One sensible way to correct this would be to define a Zone that controls the vocal processing, so we can only hope that this (or something similar) is in the pipeline for a future revision.

you can't program some Zones to receive sustain pedal or expression messages and others to ignore them... and so it goes.

Despite all of its shortcomings, playing the Grand can still be quite rewarding. It's relatively straightforward to insert the presets you want into a given Sound then tweak them to taste. Having done so, the eight Zone buttons mute and unmute the first eight Zones so you can activate or defeat the presets at the touch of a button, whether to add layers or to replace one sound with another as you play. (Unfortunately, there's no Shift key that might allow you to control all 16 Zones at any given time.) Behind these, eight knobs provide coarse control over the global EQ as well as the time constants and levels of both of the master effects. Consequently, I could see the Grand being wheeled out at festivals for bands that need pianos with (maybe) a bit of strings or something of that ilk, but it faces serious competition in that arena.

Finally, there's the question of the price. At around £2000, the Grand occupies the same range as instruments such as the Roland RD-2000 and Yamaha CP88.

But the price comparison that may be the most telling is with Kurzweil's own K2700 workstation, which, for not much more wonga, offers much more of... well, everything. Capable of creating all of the Grand's sounds and a gazillion others, the K2700 is vastly (on this occasion, pun intended) better value so, as far as I can see, the only justification for the Grand is its simplicity. Whether you view that as a boon or a shortcoming is, of course, up to you.

Conclusions

I have huge affection for Kurzweils and, even today, I use my K1000 racks and K2500 for some sounds that are simply nicer than the equivalents produced by my other synths. But, after a few weeks with the Grand, I still couldn't think what it would replace or where it would sit in either my studio or my live rigs. So why does it exist? I can only conclude that the powers-that-be at Kurzweil believe that there's an untapped market of (primarily piano) players who don't want to edit in any depth and simply want access to a library of bread and butter Kurzweil sounds. They may be right, but I think that they will have to sort out a number of shortcomings and teething problems if it's to be a success.

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READING THE ROOM

Producing With Empathy

Creating an environment where artists can flourish means building trust, understanding people and valuing everyone's input.

STEPH MARZIANO

When I first started working in studios, I found it incredibly intimidating. More often than not I was surrounded by men who were confident and sure of their ideas, at a time when I was unsure of my own voice as a producer. I soon learned that the loudest person in the room often got their point across the quickest and I remember thinking: Where do I fit in here? Where does the shy or quiet artist fit in here? I didn't want to become a bully just to make myself heard, I wanted to be the facilitator for artists whose voices had been overlooked. When I started producing with empathy it all changed.

A lot of the artists I work with tend to be women and non-binary people who have had negative studio experiences with (predominantly male) producers in the past. I once worked with a musician who had had such a bad time with her last producer that she quit music for five years. At first she struggled to even look at me. I knew I had to take my time and allow her the space to find her conviction in the studio again. At that time I felt that this was more important than how the recordings would turn out. But, with our creative trust building she became more confident and, as a result, more able to express her vulnerability. Within a few days she was thriving. It's easy to get bogged down in recording efficiently and keep one eye on the clock, especially with time and budget constraints, but I truly believe that we wouldn't have made the amazing record we made without spending the time to build our personal relationship first.

Forming a strong bond with the artist is vital in order to win their trust, obtain the best possible performance and make sessions enjoyable! This photo shows the author, centre, with the four members of Picture Parlour: from left, bassist Sian Lynch, guitarist Ella Risi, singer Katherine Parlour and drummer Michael Nash.

So here are my tips to producing with empathy.

Be In The Artist's World

I believe that most great songs come from a place of vulnerability, whether you're working with a brand-new artist or a veteran songwriter. Therefore, every interaction you have with an artist about their song matters. By actively listening and engaging with them you, as the producer, can create an environment where they feel both supported and encouraged. This empathetic approach helps establish trust and fosters a collaborative atmosphere that allows everyone's creativity to flourish. So, before setting up any microphones or opening up Pro Tools, it is vital to

understand the headspace that created the song. This is the main reason why pre-production is so important.

It's crucial to me to begin each production in a social setting, so that I can get to know the artist and find out what makes them tick. I tend to meet them for a coffee so we can have a deep dive into their emotional state, what's happening in their lives and their motivation behind writing. After initial meetings I then like to make a shared reference playlist and constantly discuss the artist's song choices. It keeps the creative ideas flowing before you even hit the studio, and helps you to get to know their musical world. By the time we're close to recording, the relationship is already formed and the trust is there.

This is my way of creating a common understanding that serves as a platform for discussions about structural changes and the sonic palette of the production. When I'm making these suggestions, I try to phrase them as questions and never as a "you should/shouldn't" statement. I can be confident in my ideas while making space for the songwriter to feel valued and understood. The key thing to remember is how personal this song is to the musician. If the story element of the song is the most important, how can I facilitate and enhance that? After all, it's our job as record producers to understand both the artist and their audience.

No Hierarchy

My most important way of gaining trust in creative spaces is by having a 'no hierarchy' rule. No matter who is in the producer's chair, we're all in this together.

I used this approach recently when I produced a new band called Picture Parlour. They ooze '70s appeal, with lead singer Katherine delivering wonderful crooning vocals. When we recorded the bass, myself and my engineer Isabel Gracefield put blindfolds on while bassist Sian tried all the different bass guitar options. Sian played a chorus with each bass as Isabel and I sat there and listened. Then the band, engineers and I voted on the best tone. As a group, we debated and guessed which one was which.

We could have easily chosen the bass sound quickly without any of this, and it definitely would have saved time, but the group camaraderie it gave was immeasurable. The band were buzzing about the experience and Sian's next bass takes were near-perfect. After that, every time we tried a new sound I could see every person in the studio feeling hyped to be there, because they knew that they all had a voice in the room.

Part of the art of the producer is making sure everyone involved feels that their voice is being heard. Steph Marziano (left) discusses a point with the band and with engineer Isabel Gracefield (seated).

As every reader knows, working with a band is very different from working with a solo artist. It involves constantly juggling different personalities and perspectives. Working with a band is all about observing the dynamics. Because of this, I find it incredibly important to learn how the band members communicate and work together before I start to chime in. Compared with a solo artist, I try to spend even more time with a band before hitting pre-production, in order to understand the relationships within the band. Even though the songs might be written by only one or two people, the other members might have great arrangement ideas, and those could be missed if the producer doesn't make space for them. It's easy to just look to the songwriters when making decisions, but involving the whole team ensures everyone has a stake in the recording process and makes everyone's performances improve. I think of us as one big extended family, with my role being the 'fun aunt' letting people try whatever they like.

For example, I may have hired a vintage Ludwig Black Beauty or 400 snare drum, but the drummer might want to try their own. It's easy to dismiss their snare if it's not the sound I originally imagined for the record, but it only takes three minutes to try a pass with it. We'll probably collectively make the decision to use the Black Beauty, but we might use theirs for another song, or as a sample to be put over the top. It doesn't take much time or commitment, but it instantly involves the drummer in a creative decision rather than dismissing them outright. Small things like this across a whole recording session add up and definitely make a difference.

Engineer and mixer Isabel Gracefield puts it perfectly when she says: "When everyone feels heard, that's when the sonics can be pushed. If I'm feeling valued as an engineer, I'm going to do my most creative work, and that passes along to the musicians, engineers, assistants, and so on. We're all in it together."

Can't We Have Fun?

There's a reason we call it "playing music". We're supposed to have fun! Isn't that why we got into music in the first place? If we wanted to be serious, we'd all do different jobs! When you allow the musicians' thoughts to be heard without fear of judgement, you're letting them come out of their shell. Wacky ideas lead to confident playing.

When the artist and band are having fun, their energy and enthusiasm is »

» palpable and their performances have a sense of liveliness and authenticity. It fosters a relaxed atmosphere that encourages experimentation, improvisation and risk-taking, leading to captivating musical moments. Ultimately, the joy experienced during the recording process is carried through into the final track, as listeners can sense the passion and pleasure that went into its creation.

Sometimes it can take three 'wrong' ideas before you get to the great idea in the studio. Other times, an idea that you initially think won't work can end up being the thing that glues the whole song together. A wacky snare sound can inspire another band member to think of a new percussion part, or to have the middle eight of the song sound like it's in another room. If we don't let musicians try these things out we could be missing out on endless opportunities, so it's important to allow space for these moments. Inclusion fosters a sense of ownership and investment in the song or album, leading to stronger performances and a deeper sense of cohesion. By acknowledging and embracing the diverse perspectives within the band line-up, we as producers can cultivate an open space where everyone feels respected and valued.

When I was a studio assistant I was always fascinated by how producers would record vocals with their artists. Everyone did it a different way! One of my favourite producers would kick everyone out of the studio besides the vocalist and himself. The intimacy of the takes it gave was incredible — so I tried the same thing with Katharine from Picture Parlour. The lyrics were personal, and I assumed it would be intimidating to sing in front of an audience. After six full takes, the vulnerability of the verses was perfect. Katharine had felt every single emotion while singing. Her lyric "I'm not sure I know my body yet" was cutting and impactful. But something in the final chorus was lacking. I had forgotten how much of a frontperson Katherine is.

We got the whole band and friends in the live room and Katharine gave a full performance in front of us all. I handed her my sunglasses, as in the scene from Adam Sandler's *Big Daddy*, and the vocal take was spectacular. I've never seen someone develop such an aura in the studio so quickly. Every artist and musician is different, and I truly believe the most important part of the job of producer is to understand how they work in order to enhance and facilitate their individual qualities in the way that makes the most impact on the audience.

Singer Katherine Parlour dons the sunglasses to deliver a vocal take that is a true performance.

Recently when recording extra backing vocals, I've got into the habit of putting a waterproof microphone in a metal cup of water and having the artist sing into it. I put a lot of effects on it and let the singer do whatever they want. We call it the "fun take". Amazing ad libs come out of it, but most of all, the singer comes back into the control room feeling incredible. By understanding what the song means to the artist, we can cultivate an environment that allows them to feel every emotion surrounding it — and that includes joy as well as the darker or more intense emotions.

Managing Time & Budget

Prioritising empathy in the studio environment creates an ideal atmosphere for fostering artistry. However, it is crucial to acknowledge the limitations of low budgets and time constraints. It is totally possible to balance empathy and efficiency in the studio, but the key lies in meticulous organisation and scheduling. By allocating ample time for pre-production, we ensure that we have the necessary resources and flexibility to explore creative possibilities during recording sessions. Guide tracks can be prepared in advance, or recorded at home (as the 'demo'), allowing us to optimise our time for more experimental endeavours. When I'm using bigger studios, I tend to use that time to let the artist let loose on the ideas that we can only really try in those spaces, even if it might mean that we end up using my smaller studio to track vocals or to edit. I also save time with my theory that the better the artist feels, the better their performance. We therefore don't need to record a million takes of each part. My biggest advice for producing with empathy is to develop these time-management skills so that when the creative juices are flowing, you've already put the work in, and know that you have the time to explore and have fun. By developing a solid musical and emotional relationship with the artist before setting foot in the studio you've established the space where creativity can flourish.

I used to think I wasn't strong enough for this job. Internalised misogyny taught me that I didn't have a place making big bombastic records. If only I had known that quiet confidence and care was what a lot of artists need. After years of producing, maybe now I am one of the noisier ones in the room. But if I'm loud, it's with encouragement. And an extra pair of sunglasses.

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Focal ST6 Solo6 & Sub12

Active Monitors

Focal's new active nearfield speakers represent a significant update on their predecessors, and the matching subwoofer is a powerful beast!

BOB THOMAS

French speaker manufacturers Focal first came to prominence in the hi-fi market, where they remain a major player. Since 1982, they have also been building a stellar reputation in the studio monitor market. Their newest range is the ST6 series, which comprises the Twin6 (reviewed in *SOS* October 2022) and the subjects of this review, the Solo6 and Sub12.

The ST6 Twin6 and ST6 Solo6 take over from the (non-ST) Twin6 Be and Solo6 Be in Focal's portfolio, whilst the Sub12, with its 13-inch driver, assumes the title of Focal's largest ever (and heaviest) active subwoofer from the SM6 series' 11-inch Sub6.

At first glance, you'd be forgiven for thinking that little has changed since between the Solo6 Be and the new ST6 version. Although I noticed that the ash-veneered side panels are now stained a slightly more vibrant shade of red, I completely missed the fact that the Solo6 is approximately 0.5cm larger all round and 2kg heavier, thanks to its thicker (22mm) panels and revised internal braces. Other obvious changes are the bevelled side panels and the front baffle's rounded top and bottom edges, aimed at reducing cabinet diffraction. The ST6 Solo6 features a taller version of its predecessor's reflex port. This features internal and external flaring designed to create a laminar air flow, which helps minimise distortion, enhance the bass response, and reduce port noise.

The Focal logo has moved from the front baffle to the top of the cabinet. And, in a major departure from the Solo6 Be, Focal's signature beryllium inverted-dome tweeter now sits inside an acoustically optimised baffle and directivity guide.

Waveguide

The baffle itself appears to me to be an evolution of those found on Focal's Shape and Alpha Evo speakers. The tweeter sits in the centre of an obround

Focal ST6 Solo6 & Sub12

PROS

- Focal's signature beryllium tweeter is smooth and highly detailed.
- Responsive, wide-bandwidth, low-distortion bass/mid driver on ST6.
- Superb integration between tweeter and bass/low-mid driver.
- Sub12 is fast and detailed, and capable of both shaking foundations and delicately delineating the extreme low end.
- Seamless integration between Sub12 and Solo6.

CONS

- Solo6's Focus mode doesn't float my boat (though it might float yours).
- Sub12 is large and heavy.

SUMMARY

Separately, the Solo6 and Sub12 are superb studio monitors that deliver performances of the highest quality. Together, they are amazing.

(racetrack-shaped) recessed area, whose angled boundary walls create a somewhat 'dog-bone' shaped outline. The walls in the centre of the obround above and below the tweeter have a steeper angle than those on the curved ends, with the aim of minimising both the tweeter's interaction with the bass/mid driver, and diffraction effects arising at the top edge of the cabinet. The shallower angle of the curved end sections is said to deliver wider dispersion (and so a larger sweet spot) compared to the older Solo6 Be.

Another visually obvious change is the large grille that protects the beryllium tweeter. The grille contours are shaped in order to minimise any potential influence on the tweeter's high-frequency dispersion characteristics. Though not the first Focal studio monitors to be given a tweeter grille, the ST6 series speakers are the first to have been fitted with this contoured version.

The beryllium tweeter in the Solo6, like that of the Twin6, incorporates Focal's proprietary Infinite Acoustic Load (IAL) housing, in which the tweeter is mounted in its own enclosure, which very gently absorbs the sound from the rear of the tweeter diaphragm. The benefit of this approach is that it improves articulation and greatly reduces distortion in the 1-4 kHz range, resulting in a more natural overall sound.

W Is For Woofer

The ST6 series' 6.5-inch bass/mid driver is a new design incorporating another proprietary Focal technology, the 'W' composite cone. Made up of a structural foam core sandwiched between two sheets of woven glass tissue, this cone's combination of lightness and rigidity, together with the precise control of the cone's damping characteristic afforded by varying the thickness of the foam core, is said by Focal to deliver very low distortion and an excellent phase response.

The cone has a deeper profile than that of its predecessor, which makes it more rigid and also means it can be made 1mm thinner. The voice coil is deeper (20mm) and more powerful, yielding 40 percent greater excursion, and the central dust cap is made from a lighter cardboard than before. All these enhancements not only make the new driver more responsive than the Solo6 Be, but also allow it to deliver a wider dynamic range and frequency response. And, of course, the driver also incorporates Focal's patented Tuned Mass Damper surround, described in detail in our review of the Twin6.

The ST6 Solo6 is an all-analogue speaker, so there are no digital inputs, and all response adjustment takes place in the analogue domain.

BASH Street Kids

The Solo6 uses the same amplifier topologies as the Solo6 Be: a BASH (Bridged Amplifier Switching Hybrid) amp powering its bass/mid driver and a Class-A/B amp driving the tweeter. The new speaker's amplifiers are rated at approximately half the power of the Solo6 Be's (80W RMS BASH and 50W RMS Class-A/B), although at 110dB, their maximum SPL is only 3dB lower. »

» Unlike some competitors, Focal remain entirely analogue in their approach, so you will find no digital inputs, D-A converters or onboard DSP on the Solo6. Instead, a -10dBu/+4dBV input level select switch feeds a simple EQ setup. This offers ±3dB shelving LF and HF EQ controls, at 250Hz and 4.5kHz respectively, to help compensate for room anomalies, and a 160Hz midrange peaking band to help deal with console reflections. A switched high-pass filter acting at 45/60/90 Hz (or off) allows you to set the optimum crossover point when using a sub without its own crossover. Last but not least, Focal's unique footswitchable Focus mode disengages the crossover and tweeter and sets the bass/mid driver's frequency range to 100Hz-10kHz, allowing you to check a mix through Focal's approximation of a full-range, single-driver loudspeaker. Switching requires either a normal latching or momentary guitar pedal-style footswitch, and can be chained between multiple ST6-series loudspeakers in an installation. An LED mounted in the tweeter baffle next to the power LED illuminates when Focus mode is active.

The Lowdown

Compared to the Solo6, the Sub12 is an absolute monster, both in size and weight. Its cabinet, which contains a 13-inch driver, amplification and controls, weighs in at a total of 62kg and encloses a volume almost seven times that of a single Solo6. The cone of the Sub12's driver is again of a W composite composition, delivering a frequency response of 28-400 Hz at a maximum of 124.5dB SPL, thanks to its 600W RMS BASH amplifier. On the rear panel, input sensitivity can be switched between -10dBV and +4dBu, and signal polarity can be inverted, if necessary, prior to the fully variable 0-180-degree phase control. Continuous 50-150 Hz low-pass filter and -24dB to 0dB level controls are followed by a switched 45/60/90 Hz (or off) high-pass filter. The LFE (Low Frequency Effects) and left/right audio inputs, like on the Solo6, are all analogue affairs, as are their associated outputs. The left and right outputs are high-pass filtered at the frequency set on the Sub12's filter.

Alongside the two quarter-inch TS I/O jack sockets for the Focus mode switching sits a second pair that allow footswitching and chaining of the Sub12's 2.1 bypass function. This mutes the Sub12 and defeats its high-pass filter, feeding full-range signals to the left and right loudspeakers, allowing quick and easy comparisons between the 2.1 system (with sub) and a normal stereo setup (without sub).

At 62kg and measuring 60 x 48.7 x 56.8cm, the Sub12 is something of a beast!

Setting Up

Unless you have plenty of space behind your mixing console or studio furniture stand, setting up a Solo6/Sub12 2.1 monitor system in a conventional fashion does pose the question of where to put the Sub12, since it takes up a significant amount of floor space. After a fair amount of experimentation in my listening room, I settled on positioning the Sub12 so that its driver was centred between, and in the same vertical and horizontal planes as, the Solo6's bass/mid drivers.

Balancing the 2.1 system wasn't particularly difficult. After Focal's recommended 20-hour run-in period and some careful listening, I settled on high-passing the Solo6s at the Sub12's 60Hz switch position, and low-passing the Sub12 by ear at approximately the same point. I also attenuated the Sub12 by approximately 10dB. Having done all that, I engaged the 2.1 bypass mode on the sub and started listening to the Solo6 speakers in their full-range mode.

In Use

When I first listened to the Solo6 they seemed very slightly more forward-sounding than I'd expected. However, once the system was set up and I started listening critically, I came to realise that it was the level of transient detail and clarity overall that gave me that initial impression.

The detail was especially noticeable in one of my go-to audition Hybrid SACD discs, the album *Spes* by the Norwegian all-women choir Cantus (whose singing you may know from *Frozen* and *Frozen II*), with Frode Fjellheim (voice and synthesizers) and Snorre Bjerck (percussion). Originally recorded in DXD 352.8kHz/24-bit format, in a Norwegian church with superb acoustics, *Spes* is a CD with many subtleties that demand much of a loudspeaker, but which also afford opportunities for a great speaker to shine. The Solo6 gave a superb account of itself in stereo playback. The sound stage was wide and deep, the centre image rock-solid, vocals from the choir and the synthesizers and percussion were rendered impeccably, and natural reverberations fading into the dark silence of the cathedral ceiling — the sternest test of all — retained a high level of detail.

I heard the same level of solidity, width, depth and detail in another favourite, *Via Crucis* by Christina Pluhar and L'Arpeggiatia, an album of early Baroque and traditional Italian and Corsican music. This is another recording that makes enormous demands of a loudspeaker's ability to resolve detail, and the Solo6 responded in exemplary fashion.

Having gone through stacks of my favourite CDs, and a few of my own recordings, I came to the conclusion that the Focal Solo6 has that rare ability to be both highly accurate and unforgiving when I was in 'critical listening' mode, and enjoyable to listen to when all that I wanted to hear was the music.

To my ears, the combination of the Solo6's beryllium IAL tweeter and its Class-A/B amplification delivers a smooth and extremely detailed performance that

The Sub12's integration options include high- and low-pass filtering, phase adjustment and switchable polarity.

has a natural quality and isn't a tiring listen. The new bass/mid driver and BASH amplification form another very impressive combination, whose increased excursion and control — coupled with the reductions in low-end and midrange distortion delivered by Focal's patented TMD surround — delivered a powerful, articulate bass and an impressively well defined and dynamic monitoring performance overall.

The integration between the Solo6's tweeter and its bass/mid driver is exemplary and results in a wide, deep and detailed sound stage with a solid centre image and precisely positioned instruments and voices. If you're working in genres that depend on extreme low-frequency detail you'll need to add a Sub12, but if you're not, the Solo6's low-frequency performance is more than capable of providing enough information in the bass to enable you to make good mix decisions.

However, all gods have feet of clay and, for me, the clay in the Solo6 lies in its Focus mode, which didn't really work for me. I really liked the Focus mode in the SM9 and Trio11 Be when I reviewed those, as their switch from three-way to two-way operation delivered practical benefits and added to their useability. My personal opinion is that a Focus mode with a frequency response curve similar to that of an Auratone or a Yamaha NS-10M would have been more useful — but your mileage may vary.

Sub12

To put the Sub12 through its paces I used Deadmau5's Grammy-winning CD *4x4=12*. This EDM classic is a particularly tough test of a loudspeaker's ability to resolve and reproduce detail in the sub-bass region. Simply put, the Sub12 produced the fastest, most detailed and most coherent low end that I have heard to date. My poor neighbours must have thought that a rave had started up in their back yard.

But the Sub12 isn't only about EDM synth basses shaking the local foundations to the core. It's also about reproducing low frequencies as part of 2.1, surround and Atmos systems. In the 2.1 review setup, the Sub12 produced exceptional results, integrating perfectly with the Solo6 pair and seamlessly extending their low-frequency response.

Overall

With its relatively compact dimensions and rear-panel mounting points, the Focal Solo6 is capable of fitting into any recording or playback environment, be it as a pair of shelf, console or wall-mounted nearfield monitors in a professional or project studio or a mobile truck, or as part of a multi-speaker system. Combining a pair of Solo6s with its partner Sub12 results in a 2.1 nearfield studio monitor system of the highest quality, the only constraint on which is the space required for the Sub12.

On their own, a pair of Solo6s are a superb nearfield monitoring solution at an attractive price given their performance level. Add in a Sub12 and you've got a 2.1 nearfield monitoring system that is capable of resolving the finer details in the extreme low frequencies from both acoustic and electronic instrumentation and percussion. If you have the need, budget and space for a system with the capabilities and performance of Focal's ST6 series Solo6 and Sub12 combination then I urge you to audition them. You won't be disappointed.

ALTERNATIVES

If you're looking for a pair of nearfields, consider the likes of **ADAM Audio**, **Dynaudio**, **Eve Audio**, **Genelec**, **IK Multimedia** and **Neumann**. There are fewer competing options for the Sub12, but if space is an issue, **Genelec**, **HEDD Audio** and **Neumann** offer more compact alternatives.

£ Solo6 ST6 £2500 per pair. Sub12 £2250 per speaker. Prices include VAT.
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Women in the studio world are disproportionately affected by issues around employment rights.

SAFER SPACES

Workers' Rights In The Studio

Up-and-coming engineers and producers deserve protection from exploitation, discrimination and sexual harassment.

OLGA FITZROY

In this issue we celebrate the success of women at the top of their audio game. At the same time, I'm conscious that many would-be colleagues have found the barriers just too big to overcome, and many are still struggling to get their invoice paid or simply trying to get through a recording session without their client invading their personal space or worse.

Celebrating success does not mean that the barriers to women's representation have melted away. Indeed, the fact that it was even possible to get most of us onto one magazine cover shows that there is still work to be done. In particular, employment laws in the UK and the US leave much to be desired — and on both sides of the pond, there are those who would block or roll back progress.

In 2021, Democrat Senator John Manchin cast the deciding vote against a paid family leave portion of a US domestic policy package that would have given six weeks' paid maternity leave to all new mothers in the US, where there is currently no entitlement to paid leave. And while things are better on that front in the UK, self-employed families — a category that includes most engineers and producers — are at a disadvantage compared to their employed peers, with no paid leave for fathers or adopters.

Another area of concern is harassment in the workplace. Again, we have parliamentarians trying to slow progress. In the UK, the Worker Protection Bill, which would increase protections against sexual harassment by third parties (something the Musicians Union have been calling for for years) looks in danger of being shelved, following objections by Tory peers.

But what do these discussions on workers' rights have to do with women's representation in the audio industry?

Seeking Protection

Workplace rights matter for everyone, but they matter most to those with the least power. Many engineers and producers find themselves near the bottom of a service-oriented food chain that has record labels, major artists and film studios at the top. Careers are built on word of mouth, personal relationships and patronage from those above. Those at the beginning of their career have even less power, and are likely to have less of a financial cushion to support them if they choose to leave workplaces that are treating them badly. Women working in audio are often the only woman in their workplace, are likely to be paid less than men doing equivalent roles, and are overwhelmingly more likely than men to be the victims of sexual assault and harassment as well as suffering pregnancy and maternity discrimination. These issues are amplified for those with intersecting characteristics of race and disability.

One of my last projects before stepping back from the board of the Music Producers Guild (MPG) last year was writing a submission to the UK Parliament's

Women and Equalities Select Committee for their inquiry on Misogyny in Music. The stats made for sobering reading: 90 percent of women reported experiencing discrimination because of their sex or gender, 90 percent had experienced online harassment by others working in the industry, 53 percent had experienced sexual harassment at work, and 21 percent had experienced sexual assault.

Gold Standards

While we wait for lawmakers to get their act together, many in the industry have been working on other initiatives to make workplaces safer. In the UK, work continues to set up an Independent Standards Authority for all creative industries, CIISA, which will cover film, TV and music. This will have the power to independently adjudicate and enforce judgements in workplace disputes. If implemented well, this will eliminate some of the power dynamics that occur when powerful industry figures are accused of wrongdoing, and give victim-survivors the confidence to come forward, as well as providing a fair process to all involved.

There are other recent developments born out of necessity during Covid which also make the industry more accessible to people from all different backgrounds. Remote working for mixing is now the norm, allowing those with caring responsibilities or living outside of the main industry hubs to compete on a level playing field. In London, triple orchestral sessions were stopped during Covid, and the larger studios are sticking with two-session days, giving a more humane working day for studio staff and musicians. While engineers' and musicians' levels of concentration and professionalism undoubtedly delivered on those long triple-session days, quality and creativity will be even better when working days are kept below 12 hours, not to mention the health and wellbeing of all involved.

The numbers of women and other underrepresented groups in audio workplaces keep on growing, and I am hopeful that this in itself will create safer and more supportive workplaces for all. While the negative aspects of our online lives are well-documented, there are also positives. Audio forums can no longer operate under sexist names disguised as 'banter', and women and other minorities have formed online communities providing everything from networking to technical advice, 2% Rising and SoundGirls being some of the best known.

Let us keep celebrating the wins, and using our creativity not just to make great records, but also to make great workplaces.

Olga FitzRoy is an award-winning recording engineer and mixer with credits including Coldplay, The Crown *and the London 2012 Olympics. She served on the board of the Music Producers Guild from 2019-2022 and has previously stood as a parliamentary candidate for the Labour party in Croydon, South London. The views expressed in this article are those of the author.*

Photo: Cherie Blair Foundation for Women and CARE International UK

Intellijel Cascadia

Semi-modular Synthesizer

Intellijel pack their considerable Eurorack experience into a single standalone synth.

WILLIAM STOKES

"We ended up getting four times the orders we were expecting, so we have been scrambling like crazy to get Cascadias built," Intellijel's Danjel van Tijn confessed to me before sending over a Cascadia for review. It's no surprise: the Canadian Eurorack veterans' debut desktop instrument has been the subject of much anticipation this year, and with a litany of outstanding Eurorack modules and cases behind them it's safe to say that fans far and wide (or should that be, the Intellijentsia?) have been gleefully awaiting the Cascadia, a sleek and elegant instrument constituting a smorgasbord of Intellijel goodness in one desktop-friendly unit.

It's certainly high time for something like the Cascadia to make an appearance: up to now Intellijel haven't offered a complete pre-assembled system in the way that fellow Eurorack heavyweights Erica Synths and Make Noise have, nor do they have any systems constructed of existing modules melded together beneath a single faceplate, *à la* ALM/Busy Circuits' System Coupe or Endorphin.es' Shuttle System. The Cascadia has recognisable functions from across Intellijel's product line (and its layout is

redolent of a massively scaled-up Atlantis), but contains none of their modules *per se*. So, in case you were wondering, you won't find any Plonk percussion, Rainmaker delay or Tetrapad touch control here. The Cascadia is demonstratively its own thing, designed from the ground up to incorporate aspects of both East and West Coast synthesis techniques and primed for the purposes of synthesists both new and experienced.

Mod Cons

"Cascadia leverages Intellijel's long commitment to modular synthesis and coalesces it into a single package of sonic possibility," goes the introduction to the manual. "Whether used standalone or as part of a larger system, we at Intellijel believe this is the synth to bridge the past to the future; the novice to the pro; the desktop to the Eurorack." That's probably about right: it does a good job of combining an analogue signal flow with a digital brain, resulting in both punishable circuitry and USB-C compatibility (not to mention breezy firmware updates and nifty MIDI interfacing); plus the starting point of the Cascadia's layout is certainly that of a conventional Eurorack system — measuring around 68HP in modular terms, a hair over the width of Intellijel's diminutive 62HP Palette, it even has two '1U' utility rows, in true Intellijel style (the company are, after all, proprietors of their own widely adopted 1U standard).

It is, however, its semi-modular-ness that sets the Cascadia apart from the likes of the System Coupe or Shuttle System, both in terms of ambition and in terms of its accessibility. By this I mean that it has many aspects of compact, plug-in-and-play semi-modular instruments like Make Noise's 0-Coast and Pittsburgh Modular's Taiga, but at the same time seeks to cram more or less the functionality of a fully furnished 6U system into a form factor not much bigger than your laptop.

All things considered, the Cascadia strikes me as a miniature modular system with some bonus normalling under the bonnet, as opposed to a hardware synthesizer with the capacity to have its connections 'broken' or 'rerouted' — which is why this review might read more like a rundown of modules in an actual Eurorack system than as one of a single instrument.

Roll Call

From left to right, the Cascadia's top-most '1U' row of modules consists of pitch, gate and trigger inputs for VCO 1, a highly flexible effects loop which I'll come back to anon, and an Output Control section. This offers individual outputs for the wavefolder and primary VCA (aka VCA A), as well as two inputs labelled Main 1 and Main 2 which can send their summed signals to both the quarter-inch line out on the Cascadia's rear panel and the 3.5mm Main output on the top right of the panel. There's a Drive knob that comes with a highly useful switch for soft clipping, which can either tastefully add additional harmonic detail or add a fair bit of dirt to the Cascadia's main signal, relative to the position of the Drive knob.

The next, larger row down begins with a MIDI-CV interface offering eight outputs, converting pitch, velocity, clock and many more messages received at the USB or MIDI input into voltage signals for routing around the Cascadia's panel. Four buttons down the left side of these outputs perform a variety of functions such as MIDI Learn, CC output assignment, LFO division, incoming MIDI Clock division, LFO waveform and also tap tempo, which is very handy.

Next along is a line input with level control and handy two-colour metering, then the Cascadia's main mixer section. This has sliders for two patchable inputs, normalled to receive the ring modulator output and a sine wave from VCO A, as well as sliders for pulse and saw waves, a sub oscillator (with useful mode switch) and noise. Outputs for unattenuated triangle, saw and sub waves from VCO A, and noise, lie beneath. There's also a second soft clip switch, which further endows the Cascadia's output with additional harmonic resonance.

All these sound as good and characterful as you'd expect, and the relationship between the sub oscillator and soft clipping switch is a particular timbral highlight. Perhaps most impressive to me here, though, is the noise generator. A majority of the noise generators on hardware synths tend to occupy a supporting role at best, augmenting a voice with some air, perhaps, or existing just to help excite the filter resonance. Here the noise generator announces itself as a key part of the Cascadia's character, and the Intellijel team have clearly given a good deal of time and attention to it. Its switch offers white and pink modes, but there's also a middle position, named Alt. This produces a digitally generated noise source, whose mode can then be selected with the four buttons on the aforementioned MIDI interface. There's Cymbal noise, Crunch noise, Crackle noise and the suavely-named Velvet noise, all of which can cycle between three different sub-variations by continuing to tap their respective buttons. That's a total of 14 different varieties of noise on offer, so it goes without saying that the Cascadia performs brilliantly when it comes to percussion synthesis.

Next along is the Cascadia's main filter. This offers a generous eight different modes: 1-, 2- or 4-pole low-pass, 2- or 4-pole band-pass, 4-pole high-pass, 2-pole notch and even a phaser (which, at the end of the day, is a simple case of all-pass filtering and polarity inversion). It has three different FM inputs (one unipolar, two bipolar) and a Level knob, which — you guessed it — can clip the input filter if you want it to. The resonance circuit can seriously scream, yet at the other end of the spectrum there's a smoothness here that in no way left me wanting. Next along is the wavefolder, a West Coast classic, complete with bipolar modulation attenuator and an input normalled to VCO A's sine wave. Lastly in the row is VCA A, the first of Cascadia's two VCAs. This has an additional auxiliary input and its own bipolar modulation attenuator.

By now you might have noticed that there's a distinct preoccupation with all things noisy here. There's drive and clipping, noise, growling sub, wavefolding, ring modulation, sample & hold and more. It all comes together to create something rather special, rather fierce, that I've not seen in an instrument like this before. Its percussive potential alone could occupy anyone for days. The Cascadia may look like quite a clinical synthesizer, which it certainly can be if you want it to, but it simmers with aggression in equal measure, almost daring you to break its own rules and push its circuitry in a way that, in my opinion, is only really possible with true analogue. Delicious.

The second '1U' row is essentially the Cascadia's utility section. I like the way most of the Cascadia's patch points are almost 'sucked' into the middle of the panel, »

Intellijel Cascadia
£2059

PROS
- Happily capable of anything from the sweet to the downright gnarly.
- Incredible signal routing, effects incorporation and I/O.
- Sleek and compact design with great build quality.

CONS
- Occasionally counter-intuitive layout.

SUMMARY
A resounding success for Intellijel's first desktop instrument. Rich, flexible and powerful, the Cascadia is on the pricey side but it's every bit worth it.

On the back panel we find quarter-inch audio I/O, effects send/return, and the Expression output. Joining these are a trio of full-size MIDI sockets and a USB-C port.

» leaving the controls at the top and bottom free for tweaking without getting tangled up in patch cables. There's the sample & hold circuit, a slew limiter and envelope follower with adjustable curve and response time, a 'Mixuverter' with signal boosting, summing and multing possibilities, a triple LFO consisting of a master LFO and two subdivided LFOs with switchable divisions (named X, Y and Z, for some reason), a very healthy buffered mult section with inversion, bipolar-to-unipolar conversion, and an intriguing knob for Expression Level, which I'll return to shortly. Next along is a ring modulator with dual inputs, usefully normalled to VCO A and B's sine waves. Lastly, on the far right side, there's a second VCA circuit which has a baked-in secondary 4-pole low-pass filter; in this way it can behave like a Buchla-esque low-pass gate, but its signal path is switchable to include either or both of VCA B's and the secondary filter's outputs, which is a nice touch.

At the bottom left of the Cascadia's panel is the primary VCO A, which sonically delivers as well as you'd expect and is replete with editability: there's soft or hard sync, pulse-width modulation with a switch to toggle between centre-triggered or edge-triggered pulse wave characters, and there's also a handy switch for through-zero FM, complete with options for either AC or DC coupling, each of which have different depths and pitch-tracking responses. To its left is VCO B, a secondary oscillator with sine, triangle, saw and square waveform outputs, which can be set to track the pitch of VCO A and also switch to act as an additional LFO if desired.

Signed, Sealed, Delivered

The Cascadia's two envelopes, A and B, lean more toward East and West Coast designs respectively. Envelope A starts off as a conventional ADSR design (albeit with a couple of additional features, such as dynamic control), but there's also the option to integrate a Hold stage into the equation, and even to toggle how that stage responds to different gate types. An incredibly useful holistic Envelope Speed control essentially allows the slider control for all five stages to be scaled, considerably expanding the envelope range. This means that, at its extremes, Envelope A's attack time can range from 0.2 milliseconds to a whopping 60 seconds, and its decay and release times from 0.6 milliseconds to 60 seconds. That, in case it needed pointing out, is very. Slow. Indeed.

Envelope B, on the other hand, works more like a complex function generator, ostensibly reminiscent of Intellijel's Quadrax module. It has two stages, Rise and Fall, with a Hold stage able to be switched in and a Shape slider for morphing between exponential or logarithmic curves — not unlike Make Noise's Maths module, or the Slope section on their 0-Coast semi-modular. All three of these parameters are CV-controllable. Switches on the right side of this module offer three clever modes for this section, each yielding further sub-modes. Given the kind of synthesist I am, Envelope B actually appeals to me much more than Envelope A, and it's brilliant to have the choice. In Envelope mode it can be set to incorporate a Hold stage and also to cycle. In LFO mode it can be set to run free or be sync'ed to a clock source.

The intriguing LFV, or Low Frequency Vacillator, essentially produces a random voltage with adjustable rate and slew, similar to the Alternate mode on the Quadrax. Alongside the sample & hold circuit it's more or less level complete when it comes to random voltage, which I was pleased to see. Lastly there's Burst mode, which accepts triggers to output an LFO that rises or decays in amplitude over time. Once again this is highly editable — it can respond to held gates by holding the LFO amplitude high, and it can cycle that response for repeating patterns of rising or decaying pulses.

Attentive readers may have spotted that, all things considered, while the Cascadia is technically a monosynth, there's the potential for something more like a duophonic synth here, since there are enough oscillators, filters, VCAs and envelopes for two full synthesizer voices, and then some. Of course, bringing the ring modulator into play and experimenting with filter resonance patching opens up further potential for melodic and timbral dynamism, and that's all before scraping the surface of the Cascadia's I/O and effects capabilities.

Side FX

I've alluded to the Cascadia's effects loop — this is an aspect of the synth that really shines when paired with its other sound-sculpting tools. Not only does it have a conventional quarter-inch effects loop available, this can be placed anywhere you choose in the signal path thanks to the FX input and FX Mix output at the top of the panel — this could be before or after the VCA, the filter, the secondary filter, or all of the above and more. VCA A can snap shut on wild distortion, or VCA B can modulate through a delay pedal and send fluttering envelopes diffusing out into space — I got particularly good results with an EHX Memory Boy analogue delay pedal. It's things like this that really prime the Cascadia for live performance as well as studio use,

The Cascadia front panel measures 348 x 246mm and is 66mm tall from its rubber feet to the tips of its many knobs.

and in this respect I couldn't really ask for more. If you've read any of my previous reviews, you might have gathered that I'm rather sceptical of onboard effects in general, so I'm totally behind Intellijel's decision to put their energy into effects integration and not the effects themselves. The cherry on the cake when it comes to effects is the Expression Level knob and patch input in the utility row, mentioned previously. This links to a quarter-inch Expression output on the Cascadia's rear panel, meaning that any outboard effects unit with an expression pedal input can be manipulated from there on the Cascadia's panel, and also controlled with CV from anywhere. Modulate delay time with an LFO, send bursts from Envelope B to flick between octaves on a pitch-shifter — best start examining the pedals on your board for those underused expression inputs, because they might just have a new lease on life. There's a master wet/dry control for the send, and also the option to send your pedals either line or instrument level signals.

Feng Shui

If I have any niggles about the Cascadia at large, it's probably when it comes to its layout. Certain things did fight against my intuition somewhat, which is a key aspect of finding one's way around an instrument like this. VCO B lies to the left of VCO A, while Envelope B is to the right of Envelope A and VCA A is dead above VCA B, which altogether feels a tad random. I generally opt to have controller modules like sequencers and mixers at the bottom of my system and sound sources toward the top, so while it made sense to me to have the envelopes at the bottom of the Cascadia's panel, I found myself wishing the mixer was down there too for quick access, particularly in performance, with the oscillators up nearer the filters, ring modulator and wavefolder. It's also not immediately clear why there isn't a dedicated output for VCO A's sine or square waves while there are outputs for its other waves, as well as for VCO B's sine and square wave.

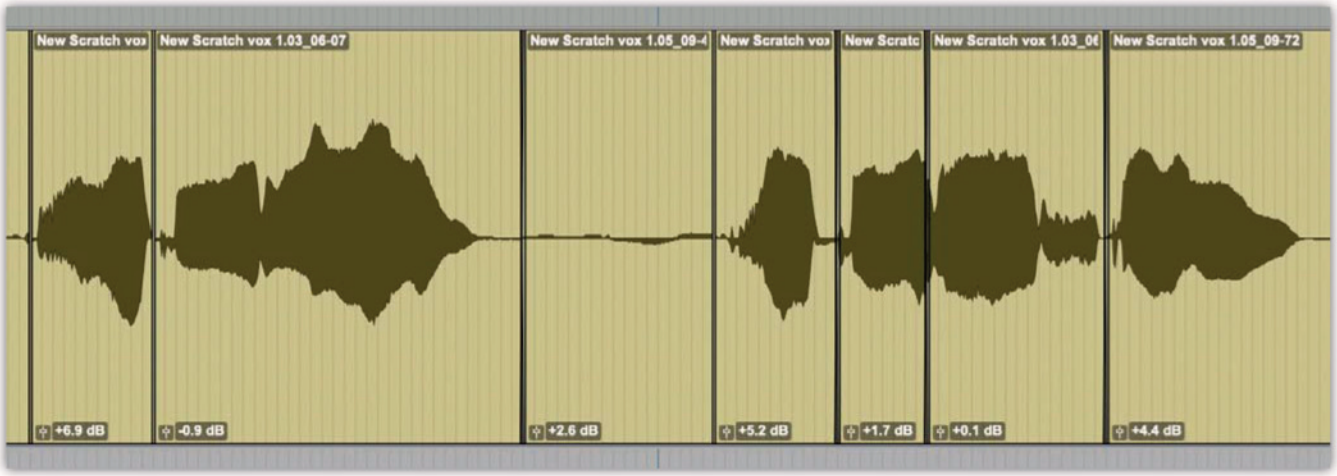
That said, I can't but return to the fact that, given the choice, this amount of functionality is more than worth a few creative workarounds to squeeze everything in, which I'm sure were needed. The Cascadia, like many instruments of its type, needs to be worked at a little — dare I say it, practiced — to yield anything close to its potential; and I'm certain that for me it will continue to yield new territory for a long, long time to come, such a playground is it.

Border Jump

Perhaps it's the sleek design influencing me, perhaps it's the effortless signal routing and integration options, perhaps it's the excellent build quality, but I can safely say the Cascadia is a synth simply oozing confidence. It's capable of such raw power and character, yet this isn't even placed at the forefront of its identity. Before getting anywhere near its wild side, it's plain to see the Cascadia is an ultra-reliable, great sounding, highly functional, highly useful miniature modular system — one that I can imagine constituting the very soul of a great deal of setups, both live and in the studio. Up to now I haven't mentioned the Intellijel Config app for Mac or Windows, which allows for an even deeper dive into the instrument's controls; particularly when it comes to MIDI, although it can also be used to change the waveform of Envelope B's Burst mode, adjust the default Alt noise type and more.

Presumably the Cascadia is named after the boundary-crossing bioregion in Western North America, which stretches from British Columbia, Canada, down to Oregon, USA; the Canadian company reckoning with those American heavyweight coastal schools of synthesis. Perhaps it comes from its many cascading mults, ready to send voltages spidering out across its panel to various destinations. I'll posit another possible reading of the name, which I find myself returning to time and again: it might start somewhere simple, but it's never long before more ideas begin to slide, tumble and ultimately cascade forth. At its price, the Cascadia is a premium instrument to be sure, but given what's on offer here and the price of an equivalent Eurorack arsenal, it's a steal.

£ £2059 including VAT.
W www.intellijel.com



Screen 1: This Pro Tools screenshot shows the typical end result of using clip gain to control levels within a vocal phrase. Gain adjustments range from -0.9dB to +6.9dB.

CLIPGAIN

The Secret To Smooth Vocal Control

Most DAWs give you the power to control the volume of recorded audio on a per-clip level — and with this power comes immense creative potential!

MIKA SELLENS

I'd like to talk about clip gain. It's a subject that, on the face of it, may seem a little dry. But clip gain has grown to become one of my favourite creative production tools, and I'd like to talk about why this is.

My music production journey began with experimental electro-acoustic sound work. Computers didn't have the processing power they have today, and dynamics had to be controlled using more rudimentary tools and effects. Much of my work involved hours of painstaking volume automation, carefully shaping a finished piece by combining pieces of recorded audio.

Fast-forward to the present day, and in the work I do with artists as a producer, engineer and mixer, something of this early sensitivity to volume levels has taken hold in my production techniques. The best example is probably my use of clip gain in vocal production.

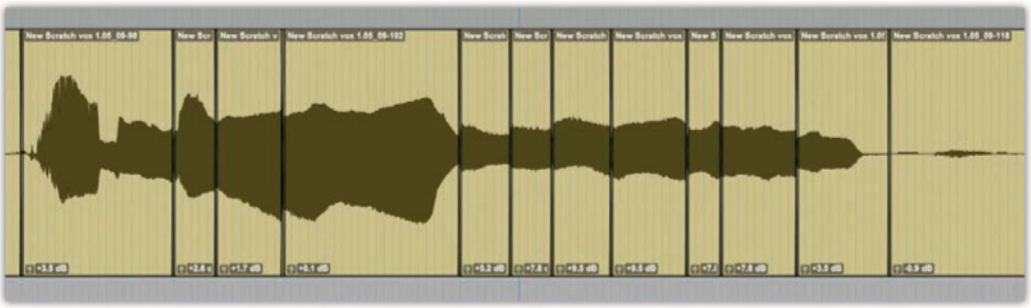
Vocal Clips

Vocals are the heart of any song. They engage the listener emotionally with the song and hence the record. I like vocals front and present and prominent in the mix, and I like them sounding smooth and utterly beautiful.

As much as I'd like every vocal take I record with an artist to be perfect (and getting a good take and capturing a great performance is certainly the starting point), most vocals require some comping and some dynamic control to make them ready for repeated listening. And it is at this point that clip gain comes in. I'll make no apologies for being a Pro Tools person through and through in my examples, but the techniques I'm going to describe are readily applicable in most DAWs.

The process begins with vocal comping: adjusting the clip gain of the takes I'm choosing for the comp as I go. And as I listen and work my way through the vocal comp, differences in intensity and dynamics begin to reveal themselves. A quick Command+E to split the audio, followed by another at the end point of the selection, and a mouse click and drag to adjust the gain of the newly created region is all I need to control the flow of the vocal. Dial it back a bit, bring it forward. Intensity and perceived volume relate to distance from the mic as well as vocal production, and adjusting the clip gain in this way can help to bring a consistency to the recording. And before you say it, I know that this is a form of manual compression. But also, I use clip gain in addition to compression for greater control.

So, as my vocal comp starts to take shape, I'll have made rudimentary clip gain adjustments during the comping process. Once it's complete, I tend to start again from the beginning of the comp and listen in more detail, making further adjustments as I go. Now we all know there are multiple



Screen 2: A single vocal phrase with a long note to end, the intensity of which is controlled via multiple clip gain adjustments.

ways to do many things in most DAWs, and that's certainly the case in Pro Tools. Rather than splitting the audio into separate clips, I could make the Clip Gain line visible and add nodes to it. But this has become the way I work. It's quick and well practised for me, and visually, it helps to bring clarity to the changes I've made.

My clip gain adjustments can be quite granular, often controlling the levels of breaths and individual syllables as well as the progression of sustained notes. Screen 1 shows two short vocal phrases, with a breath between. It is very usual for all the breaths and a good number of ending consonants to require separate clip gain adjustment to control how they land.

Once I've reached this stage, I'll likely spend some time making some tuning and EQ adjustments, followed by some compression for sound and tone, and some gentle additional levelling. This will be followed by my choice of reverb and delay for the specific vocal I'm working with.

Into The Flow

When I have the effects and sends in place, I'll render any Melodyne tuning to create new clips, then begin a second stage of using clip gain to make further adjustments. And it's at this point that the adjustments start to feel really musical, and literally like controlling an instrument, as the clip gain adjustments being made control how the audio is flowing into the effects. This gives me increased creative control over how a note hits the EQ, compression, reverb and the delay, and over how the tone of the vocal is presented. It feels as if adjusting the clip gain can smooth out and control the production of the note in the way that the actual vocalist does when producing the note. The vocalist, of course, does this by controlling the airflow through their vocal cords, and although the mic has already captured the different intensities of airflow and hence the volume progression in the note, controlling the gain of a clip almost feels like a way of adjusting this airflow after the fact. Applied this way, clip gain adjustment seems to provide an otherwise unreachable level of control over an already recorded performance.

And so to endings, and the ending flow of a long note. You might think that the example shown in Screen 2 would sound uneven, but this is yet another example where it's most important to trust your ears over your eyes. This vocal, controlled using clip gain adjustment through effects, may result in the waveform looking a little uneven, but it sounds far from it. The vocal produced is controlled and smooth in its flow. It can now sit front and centre and clear in the mix, ready for many repeated listens in the future.

So, if you've not previously explored the creative potential of clip gain, do enjoy your explorations. And if, like me, you choose to make your clip gain adjustments by chopping up the audio into shorter clips, don't forget to select all at the end of your process and Command+F to put micro fades on all of the edits you've made. Happy clip gain exploring... and prepare yourself for some seriously smooth vocal control!

Mika Sellens is a producer, writer, mixer and engineer. She is a professor of Electronic and Produced Music at Guildhall School of Music and Drama, and an Executive Director of the Music Producers Guild (MPG).

Roland AIRA Compact S-1 Tweak Synthesizer

Synthesizer

Roland's SH-101-inspired S-1 is a great-sounding synth with surprising depth.

ROBIN VINCENT

The Roland S-1 Tweak Synth is the latest in the AIRA Compact range of fun and fiddly little synth boxes. It's inspired by the classic SH-101, and promises to have the same iconic vibe, although it looks nothing like it. Roland use the same Analog Circuit Behaviour (ACB) technology in the S-1 as they do to build the plug-in instrument emulations of their legendary gear, and it also forms the basis of the Boutique range of synthesizers. So, it's probably right to say that the S-1 is more like a fun-sized version of the already petite SH-01A Boutique. However, while it has the same engine and boost to four-note polyphony as the SH-01A, the S-1 also has a few extra tricks up its sleeve.

First Impressions

I like the other boxes in the series, particularly the T-8 Beat Machine, so I had high hopes for the S-1.

The form and layout are very similar to the T-8. You've got a four-digit display, and two and a half rows of small, trimmer-style knobs. It has some buttons and the most miniature two-octave keyboard I've ever seen, and it's all wrapped up in a rechargeable plastic case no larger than a Korg Volca.

The knobs are fiddly, but that's not uncommon, so it's the sort of thing we put up with on good value, mobile little boxes. The white text on the black background is easy to see; however, there's also text on the buttons, which is far more challenging to pick out. My 52-year-old eyes are not the best, but I had to use the torch on my phone to read the text on the black keys. Holding the Shift key illuminates all the buttons in orange, but they flash if they are not enabled so this is only partly helpful. It's not terrible, I can work around it, but legibility is an important factor, and it's going to eat into my enjoyment of this little machine. However, after a couple of minutes of playing with the S-1, I was hooked; this is a delightful little synthesizer.

The Sound

The front-panel synth architecture roughly follows the SH-101/01A. It starts on the left with the modulator section featuring an LFO with rate knob and waveform

selector, followed by a combined VCO/source mixer section with a six-octave Range selector, an LFO depth control, and four level knobs controlling a pulse wave, sawtooth, sub-oscillator and noise generator. Next is the filter section with cutoff, resonance, LFO and envelope depth. Beneath that is a full ADSR envelope. The LFO has the added waveforms from the SH-01A, but other functions, such as PWM and sub-oscillator octave, must be discovered via the Shift key.

It sounds very much like a decent analogue synthesizer, and Roland have nailed that SH-101 tone, but you quickly find yourself going beyond that with the overlapping notes of polyphony and the impressive reverb and delay. The mix of waveforms is excellent, the sub fattens things out gorgeously, and the noise can widen the textures when used sparingly. The filter happily pushes into self-oscillation and can run as a slightly untuneful oscillator. Just on the surface, there's a lot to explore, especially once you've worked out how to enable the arpeggiator.

Using the two-octave keyboard is not quite as terrible as you'd imagine. I think it's a better playing experience than the touch strip of a Volca or the buttons of a Cre8audio East Beast. It somehow manages to accommodate my fat fingers, and having just over two octaves really makes a difference when bashing out a tune. There's no velocity on the keys, but then there wasn't any on the original, and you can always plug in a MIDI keyboard for more expressive control. While I was pleasantly surprised by how playable the miniature keyboard is, I would definitely recommend plugging in a proper keyboard. With bigger keys, velocity, four voices, those effects, and the space to play, the S-1 can feel like it's playing in the Minilogue leagues.

Preset sounds are folded into a system called Patterns that includes a sequence and modulation data. There are four banks of 16 sounds but only the first bank is populated. They do a great job of showing off the range of this little synth, and if you set it playing you'll find that switching Patterns is quantised to the tempo, which could be very handy in performance. Sadly, there's no song mode.

I found one anomaly — or possibly a feature — where any changes you make to a pattern by turning knobs are retained if you move to another pattern and then come back to it. I couldn't work out a way to return to the Pattern's saved sound without turning the S-1 off and on again. I found it particularly frustrating with the otherwise useful Manual button. This button sets the synth to whatever the front panel is set to, which is a brilliant feature, but you can't get back to the preset sound you were on. I eventually found a solution: the 'rLod' command, number 37 in the Menu list, which reloads the current preset. Maybe Roland could make it more accessible — this will be a bit of a theme.

Hidden Depths

The Shift button opens up some of the things that are missing from the front panel. You can set the source and depth of pulse-width modulation, change the sub-oscillator octave from minus one to minus two and make it asymmetrical. You can also change how the envelope is triggered and whether the amplifier uses the envelope or a gate. There are portamento settings, polyphony modes, level of keyboard modulation to the filter, and deeper settings for the effects.

And just when you think you've got the measure of the S-1 and you're enjoying the more-than-the-101 vibe, you discover it has a whole side-quest storyline to explore. This comes in the shape of the Draw and Chop oscillator modes.

Oscillator Draw allows you to reform the square wave oscillator into something of your own. It creates a stepped or smooth triangle waveform with 16 adjustable nodes laid out on the 16 white keys. You can push each node on the waveform up or down in height to draw in harmonics and tonal changes. Stepped mode is fizzy and aggressive, while sloped mode moves into wavetable territory. A Multiply parameter increases the waveform's playback speed, giving you a sort of cascading sync sound. There's a lot to enjoy here.

Oscillator Chop pokes holes in the waveforms. For this, you can use the square, sawtooth, sub-oscillator or noise and give them all a different Chop pattern. The effect is to bring out overtones and harmonics. You can apply a comb filter to push it into the more clangy territory.

The sound-crafting possibilities here are marvellous. Once you've mastered the tedious menu system, chopping or drawing on the waveforms can take the sound out of the SH-101 domain and into completely different places. The Comb and Multiply parameters can be accessed from the front panel via the Shift button, so you don't have to get into the menu to mess with it while you're playing. Sadly there's no opportunity to modulate via LFO or envelope, but you could use the sequencer modulation motion lanes to make per-step changes.

Sequencer

The SH-101 is famous for the immediacy and simplicity of its step sequencer. The S-1 ignores all the fun you could have had with that and does it in a more versatile but long-winded way. In step mode, you can turn steps on or off from the row of 16 white keys in banks of up to 64. You then have to laboriously hold each step and dial in the note. Not including the quick-fire step writing with ties and rests that made the 101 so funky is a missed opportunity, but the flip side is that you also get to include a bunch of other stuff while you're holding that step. You can dial in velocity, gate length, probability and a sort of sub-step ratchet effect in up to four slices.

You can also record in real time, and there's a count-in and metronome hidden away in the menu system if you can find them. With the four voices of polyphony on offer you can record chords, overdub, or jam along with your sequence.

As well as the parameters you can add per step you can also record up to eight knob movements into your sequence. So, you can add in those filter sweeps, effect plunges, waveform mixes and anything else you can think of. It makes »

Roland AIRA Compact S-1 Tweak Synthesizer

£169

PROS

- Sounds fabulous.
- Four-voice polyphony.
- Excellent Unison mode.
- Great effects.
- Decent sequencer.
- Chop and Draw waveform editing.
- Rechargeable battery power.

CONS

- Menu and sub-functions can be hard to decipher.
- Some cool features are hard to find.
- Motion sequencing prone to clashing with itself.
- Really needs an Undo button.

SUMMARY

The S-1 starts off as a miniaturised and fiddly take on the SH-101, but offers so much more that the fun of playing it far outweighs its frustrations.

There's not much to see round the back, with just a USB port and 3.5mm MIDI sockets.

» the S-1 capable of amazing changes and variations throughout a single sequence. But it's not as nuanced as it could be. Once you've recorded in a bit of motion, the S-1 will fight you if you try to adjust that same knob during playback. The new movement doesn't override what's already there and spits at you as it tries to be in two places simultaneously. You can also only clear all the motion data, either per step or for the whole Pattern, so if you get knob motion number eight wrong, you have to delete the lot. An Undo function would be really nice.

The one motion that's not recorded is, weirdly, the D-Motion. This gyroscopic technology built into the S-1 lets you control parameters by tilting and twisting the synth about the place. It's pegged as a performance device so, I guess, if you wanted something recorded into the sequence, you can just turn the appropriate knob. The D-Motion is all about picking it up and getting yourself into control.

You activate it by holding the D-Motion button, and then you have Pitch and Roll mapped to two parameters. These can be modulation, cutoff, resonance, and pitch-bend on Roll; and pan, expression, delay level and reverb level on the Pitch. It's a fun thing that some people will enjoy, and the rest of us can ignore without any loss of functionality.

Effects

Did I mention how great the effects are? The Delay can offer repeats from 1 to 740 ms, sync'ed or unsync'ed, and can go from phasing to dub and through to ambient without any bother. It's very well behaved and doesn't overwhelm or destroy itself even at full-on feedback and level. Depth is available on the front-panel knob, which switches to delay time when holding Shift.

Reverb comes with Ambience, Room, Hall 1 & 2, Plate, Spring and Modulation modes. They all sound great, and having just discovered the lovely wavering Mod one, I'm going to be using that all the time. You also get control over time, level, pre-delay, low and high cut, and density, all through the menu. Depth and time are available on the front-panel knob.

Chorus is not immediately obvious because it's buried in the menu. It's derived from Juno synths and should probably be a bit more front and centre. There are four options ranging from regular, to faster, Leslie and chilled, and they are perfectly good. There's no depth or time control; it's just there and does exactly what it should do.

Another sort of effect is the Riser function, which is hidden away in the menu system like the chorus. It's a very cool rising and falling sound that takes over the Noise oscillator. Once enabled you have a handful of different modes to add riser and downer effects to your sequences. It's fun and exciting and deserves a button on the front panel. But then many things do, and Roland have had to make a number of difficult decisions to keep to the form factor. Having those features in there, even if only accessible via the hieroglyphs of a menu system, elevates the S-1 to be much more than it seems.

Conclusion

The S-1 is a bit of a rollercoaster in terms of its feature set. The keyboard is ridiculously small, but strangely playable; the knobs are tiny, but the sound is fantastic; the sequencer is groovy, but not as fun as the original; the motion capture is exciting, but the clash of overwriting can be disastrous; it has deep levels of waveform editing, but also deep levels of menu diving.

However, as a playable, tweakable mini-synth, the S-1 has an awful lot going for it. Four voices of Roland virtual analogue in a super-portable, battery-powered form, with a sequencer you can jam with, editable waveforms and outstanding effects. There will always be compromises in something this small, but it's certainly the most interesting synth of the AIRA Compact range and it fits perfectly into that line-up. The experience of using the SH-101 is so much more than the sound; it's an instrument, an interface of per-slider functions that visually represents the sound and invites thorough synthesis engagement. The S-1 is not that, but it sounds great and stands up as a decent synth in its own right.

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Linda Perry

Songwriter & Producer

Photo: Kristin Burns

It takes talent, attitude and self-belief to succeed in the music business, and Linda Perry has plenty of all three...

TOM DOYLE

One of the best-known and most successful songwriters and producers in the industry, Linda Perry has been responsible for a string of huge international hits by the likes of Pink ('Get The Party Started'), Christina Aguilera ('Beautiful') and Gwen Stefani ('What You Waiting For?'). Perry has also worked with the likes of Alicia Keys, Ariana Grande, Courtney Love, Adele and Dolly Parton.

Before all of this, Linda Perry was an artist in her own right, most famously as the singer in San Francisco band 4 Non Blondes. She wrote their 1993 hit 'What's Up?', which reached number one in 10 countries. When the band broke up the following year, Perry embarked upon a solo career with her critically acclaimed, if commercially unsuccessful solo album, *In Flight* (1996), before she decided to instead pursue her current career on the other side of the studio glass.

Make Yourself Heard

Even in her teenage years playing in various bands, Linda Perry always had a formidable attitude. "I kept getting kicked out," she laughs. Born in Springfield, Massachusetts, she spent her teens in San Diego, before moving to San Francisco at the age of 21 where she began performing solo around clubs in the Bay Area. Spotted by 4 Non Blondes' bassist Christa Hillhouse, she was invited to join the all-female band, whose ranks also included guitarist Shaunna Hall and drummer Wanda Day.

"At first I didn't want to bring my songs into the band, because I was doing my own solo stuff," Perry remembers. "Then once I realised the band was pretty good, I started bringing my songs in, and we started getting more acknowledgement." Up to this point, Perry had had very little experience of recording. "No, at that time, I wasn't even understanding that process. I had a tape recorder. That was it, and then I ended up buying one of those four-track Tascams."

After bagging a deal with Interscope Records, 4 Non Blondes went into the studio with producer David Tickle (Split Enz, Joe Cocker) to record their debut album. It proved to be a disappointing exercise for Perry, who quickly grew frustrated that the guitar sounds she heard when standing in front of her amp weren't being accurately represented through the monitors in the control room.

"David Tickle was an interesting character," she says. "He was a great guy. He put these really great tones in front of me. Like, he introduced me to the Vox and Les Paul and that dry, in-your-face gritty sound. And I was in love with it. So, we'd be tracking these songs, and then when I'd come in and listen, I'd be like, 'What? Where's that tone? This tone sounds small, really distant, and kind of thin.' And I didn't understand at the time that it was compression and EQ'ing all the beauty out of it and putting reverb on it. I feel like David was completely off the mark of who we were as a band. And I started complaining."

4 Non Blondes' album *Bigger, Better, Faster, More* was a huge hit off the back of Perry's song 'What's Up?'.

Perry felt she was being railroaded by both the producer and the record company into releasing an album she wasn't happy with — particularly Tickle's version of the anthemic 'What's Up?', which she rightly regarded as one of her strongest songs. "We'd recorded 'What's Up?', and it was just terrible. Like, I cried. I went to the label, to [*Interscope's*] Tom Whalley. I said, 'This is fucking shit. This is not the song I wrote. Sounds like crap. I don't understand why [*Tickle*] made me change lyrics and put a solo and a marching drum and all this shit in it.' And Tom Whalley at the time was like, 'I think it sounds great.' And I'm like, 'Well, then, fuck you, too. It doesn't.'

"The album was about to be mastered. Everybody's accepting it as the end result. And I panicked. Honestly, I would have scrapped the whole album. But I knew that was going to be an impossibility. And I said, 'If I'm going to save one song, it's going to be this one.'"

Saving The Day

The singer took matters into her own hands, booking a day at The Plant in Sausalito, in an attempt to record 'What's Up?' to her satisfaction. "We had one roll of tape," she recalls. "So we only have three takes. We went in, we had a house engineer there [*Mark Hensley*], and I just started dialling in sounds with him.

"Now, mind you,' she adds, "I have no idea what I'm doing. But I hear the kick drum and I'm like, 'No, can we make it a little more warm and fatter?' Then I would move microphones, and I just did what instinctively felt like the right thing to do. Then last thing was my guitar. He dialled in my vocal and guitar with me and we recorded the song, we got the take. I knew I'd saved the song because then I heard it, and I'm like, 'Yes, this sounds like me. This sounds like us.'

"And then David Tickle showed up at something like 11 o'clock at night. He'd found out that I was up there doing that. We had to mix it because the album was being mastered the next day. And that's the version everybody heard."

Nonetheless, Perry was disgruntled when she wasn't given even a co-production credit on 'What's Up?' "I said, 'I produced this. I want credit.' And everybody pretty much laughed at me and said, 'Well, it doesn't work that way. David's deal's already done.' But I tell you, it was the greatest lesson that I got early on in my career. Because after that, that never happened again."

Out On Your Own

It was during the sessions for 4 Non Blondes' never-to-be-released second album that Linda Perry decided to go »

» solo. Work had begun with producer Dave Jerden (Talking Heads, the Rolling Stones) but the singer was suffering from creative disagreements with the rest of the band.

"Dave is amazing," she says, "but I was in a different place. The 4 Non Blondes experience for me was a dark experience. So that's where I was at. I started writing songs, and the band kept on saying, 'Well, this sounds more like something you would do by yourself.' They kept saying that to pretty much almost every song. Then I would disappear for like five minutes, come back with a stupid song. And they'd be like, 'Yes, this is great.' And then we would record that."

It was in fact Jerden who emboldened Perry to break out on her own. "One day, I was in the studio, playing piano and Dave walked in. And he was such a kind man to me. He's like, 'Listen, you're one of the most talented people I've ever met. Don't let people make you second guess yourself. You know what you want. Stick to it.'

"After he said that, we went on a little break, before we were going to come back and start doing overdubs on the record. And that's when I decided, 'You know what? I'm leaving. I'm out.'"

Interscope Records retained Perry as a solo artist, and put her back into the studio with Bill Bottrell, who at the time had recently co-produced hits with Michael Jackson ('Black Or White', 'Who Is It') and produced Sheryl Crow's multi-platinum 1993 debut album, *Tuesday Night Music Club*.

"Bill was so confident," Perry says. "He was like, 'You know, I can tell you everything I do, I can give you the same players. But you're never going to repeat what I do because you don't have my ears.' So his whole concept was, 'Yeah, sure, here, learn what you want. I'll give you all my secrets, I'll give you all my tricks, but you'll never ever be me.'

"So that was really wonderful to learn, because I did have my own ear. I asked him questions all the time and he would just put me in front of the board and he just showed me, 'This is your master volume. These are your sends. This is your EQ: high, mids, lows. This is how you cut the lows. This is how you cut the highs. Over here are compressors. These over here are EQs. These are your effects. Turn the knobs until it sounds good to your ear.'

"That's what I've been doing since then. That was like the birth of being actually the engineer."

Linda Perry's breakthrough hit as a songwriter was 'Get The Party Started', written very quickly as she tested out her new Akai MPC.

Voyage Of Discovery

At this point, Linda Perry began amassing her own studio gear, starting off with a Neumann U67 microphone, a Fairchild 670 compressor-limiter and a Pultec EQ. "I started buying gear and not knowing what it did," she says. 'I just bought it, and it sat in my warehouse. And then I just started slowly putting it all together by myself, and then had a friend come and do all the soldering. But I learned how to use it all on my own, with no teaching, no anything.

"Then I just started recording bands in San Francisco. Y'know, 'Hey, I'll record you if you let me record you. I need the practice.' And so that's kind of how I started."

Perry moved to Los Angeles in the late '90s and expanded her setup, having discovered the Akai MPC2000 sampler and Korg Triton workstation synth. "Well, it was not a discovery I wanted to make," she chuckles. "It was me just simply trying to figure out what all the weird, bad sounds were on the radio that I was hearing. Because it all just sounded so... no heart, no nothing.

"I was having a hard time responding to what I was hearing. So I wanted to find out what it was. I asked a friend, 'What are all these sounds that are on the radio right now?' And they laughed, and they were like, 'Oh, yeah, MPCs. Those are drum samples that are put on a drive. You can make your own samples and put them on a floppy disk.' I was like, 'Oh, interesting.' And then the Triton was this keyboard that had all this stuff in there. And, y'know, people were using [*Tascam*] DA-88s. I wasn't ready to go to computers yet.

"So I just bought all that stuff and started figuring out what it did. And then it was so funny, because as soon as I started hearing some of the sounds as I scrolled, I could place almost every song. Y'know, like, 'Oh, OK. This is that track? Oh, I get it.' It was super fun to understand.

"I got it all hooked up. And now I'm just kind of running through sounds. Then I was like, 'All right, well, let's just dick around!'"

Party Time

Incredibly, Perry's first musical experiment with her new equipment produced her debut hit for another artist: Pink's 'Get The Party Started', released in 2001. "I didn't have Pro Tools or a computer, so everything was just being played live," she stresses. "So I get a tempo... 120... that feels normal, average, and »

> Linda Perry: "I've never written another song that sounds like 'Get The Party Started' or 'Beautiful' ever again. If you want something original, I'm the person you come to. But if you want a soundalike, go somewhere else."

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» I lay a click down for four minutes. Then I started creating this beat on the MPC, played all the way down. Thank God I'm a metronome with time. And so I just play kick and snare, adding real shakers, bongos. I'm just going, 'What the fuck, whatever. I don't care, I'll play bongos.'

"Then I couldn't find a bass sound I liked, so I picked up my Fender, popped it through my Ampeg B-15, and I just started playing. 'OK, just super simple. Don't even think... dumb it down, Linda.' Then I lay down a wah guitar. I go to the Roland sampler, and that's got a bunch of cool sounds. There's a Clavinet and I put these stabby horns in.

"The track is coming along. I'm kind of laughing at this point. I had a [*Shure 520DX*] bullet microphone next to me because I had attempted at one point to put harmonica in there, but it didn't work. So I grabbed the bullet microphone, and I was like, 'OK, let's put a vocal on here.' And I just started thinking of every cliché I can imagine. And I just went right into it. I just pretty much made up 87 percent of the lyrics on the fly and then went back and filled in the blanks.

"All this took anywhere from 15 to 30 minutes. It was so quick. After I was done, I just started laughing, going, 'Holy shit. I just wrote a fucking pop song.'"

Thrilled and surprised by what she'd spontaneously created, Perry sent the demo to Madonna's manager, Guy Oseary. "Honestly, I don't even know if Madonna heard it or not," she says. "Then he reached back out to me and said, 'Oh, we're gonna pass on this.' And I'm like, 'OK, no problem.'

"Then, about a week later, that's when I got the weird random phone call from Pink. I called her back and I said, 'I think you have the wrong person. I'm not hip at all. And I don't know anything about your style of music.' Because she was doing that white girl R&B stuff. I was just like, 'That's not my thing.' She's like, 'Are you not Linda Perry from 4 Non Blondes that sang "Dear Mr President?"' I'm like, 'Yep, that's me.' She's like, 'I have the right person.'"

Insecure Power Girl

Pink duly recorded 'Get The Party Started' and it became an enormous international hit. Linda Perry's writing/producing career was off and running. Her next big hit was Christina Aguilera's 'Beautiful', an anthemic ballad dealing with insecurity and self-empowerment. Aguilera's vocal on 'Beautiful' was in fact the demo take that she had recorded in Perry's home studio. "During a time where everybody was getting these perfected vocals and tuning the hell out everything," Perry says, "that song was live. That demo vocal is what I recorded at my studio with a U47 through a 1073 and LA-2A.

"Christina was super-nervous that day. She brought a friend and basically when the piano started, she looked at the friend, and said 'Don't look at me.' And so I already knew, 'Oh, OK. She feels the same way as I do. Y'know, she's insecure. She's an insecure power girl.'"

Perry took the home demo to Conway Recording Studios in Hollywood and built a simple band arrangement around it. "'Beautiful' was five microphones on the drums," she says. "It was live to click, but not lined up. Bass, piano, guitar, a [*string*] quartet, a Mellotron and vocals. It was 16 tracks, that song, and everybody played to Christina's vocal."

Perhaps inevitably, once the track had taken shape, Aguilera decided that she could sing the song better. "She's like, 'Um, can I re-sing it?' I'm like, 'Oh, no, no, no. You're not touching this vocal, I know exactly what you're gonna do.' She's like, 'I know it better now.' I said, 'I know you know it better now. And that's why you're going to ruin it.'

"So the record is almost done... finalising... about to start mixing. She's like, 'Please, can I just try to sing it one more time?' I'm like, 'OK.' We go into Conway, I get her all set up. I got the gear there, the right microphone, everything. And I push record and, I mean, I'm into like, 15 seconds, I'm not kidding you, and she's already [*imitating Aguilera's over-emoting*] 'Heeeeeeey-ah-ah-ah.'

"I just stopped: 'What the fuck, dude? You're ruining it.' Then she was like, 'You're right. Can we at least punch in a couple of words?' I'm like, 'Absolutely.' So we punched in some words that I was going to do anyways, that were out of tune, because I refuse to use Auto-Tune as well. And then we came up with a different melody for the bridge and she re-sang that first line in the bridge. Then that was it. That was the vocal."

In Touch

From this point on, Linda Perry continued to build her reputation within the record industry, particularly for working with female artists. "I think that women weren't used to, or even understanding, that was a possibility," she points out. "I remember when Gwen Stefani came in, and she was like, 'Oh my God, it's such a different energy working with a woman.' And it is. It feels safer. You don't have some douchebag dude hitting on you because they think they're powerful."

Describing herself as a "boutique songwriter", Perry says she relies upon her instincts and emotions when approaching a new composition. "I don't have big, grandiose thoughts in my head when I'm writing a song," she stresses. "I don't have a preconceived idea: 'Y'know, today, I'm gonna write a big, powerful anthem.' I don't do that. I write from going with whatever my feelings are. It's being in touch with your feelings and where you're at emotionally in life. I just sit down, and I play and the song just shows up."

Old Mics Rule

Working with an array of female singers, Linda Perry has developed different vocal chains tailored to each performer. "Sometimes a chain that I think is gonna work great on somebody, because I love it, doesn't work on them at all. I've been through so many different microphones and so many different people.

"On Gwen, on just one song, 'What You Waiting For?', I used a [*Shure*] SM7, a [*Neumann*] U47, an 87 and a [*Telefunken ELA M*] 251. I used all those microphones on her vocal to get, like, all the different characters. Because the beautiful thing about microphones and gear and all of it is it's all different characters. Except for, honestly, and I'm not being a snob when I say this, but new microphones are just new microphones. You can put five new microphones next to each other and they all sound the same.

'But you know, a U47 compared to an 87: completely different. An SM7 to 251: completely different. Right now I was working with this artist, and I had her on a U47 and through an API, and I was like, 'Oh God, this technically should just sound right for her.' But then I just took it off. And I just put up an SM7 through my Neve, with very, very little compression and then I was like, 'Oh, there you are.'

"So I love that discovery. I've used [*Electro-Voice*] RE20s on people. I've used a [*RCA*] 44 ribbon mic on people. It depends on the song too and what you're doing. If you want top, you're not going to use a 44. You're going to use something a little crisper, like a 251 or a C12 or whatever. But if you want unique, you can go anywhere you want."

Perry has a reputation for being open and honest with her various star collaborators, even if sometimes they might not want to hear her characteristically strong opinions. "What I have that is the most important to me is respect," she says. "I'm someone that you can trust, and I'm always going to be honest. And people come to me all the time for my honesty and ask me, 'Hey, what do you think of this?', and they're prepared for a very raw answer. And for that, I'm so grateful."

Beyond 9 To 5

More recently, Linda Perry had the opportunity to collaborate with country legend Dolly Parton on the soundtrack to 2018 film, *Dumplin'*. The album featured Perry's reworkings of six of Parton's previous recordings, including a string-backed version of 'Jolene', alongside five new Parton/Perry co-writes.

"To me," Perry laughs, "walking in a room with Dolly Parton was just life calling and saying, 'You got to up your game. You got to take it up a notch.' Because that woman, she shines. An incredible woman and so powerful and extremely talented. So I rose to the occasion... I matched her confidence and walked into that room with her, and I cranked out six songs with her in two days. And we got along amazingly. She's a very good friend and she's wise beyond belief."

Parton's only concern prior to the sessions was that Perry would take her inherent country sound and turn it into pop. "I had a band come in, and not even a country band. When she heard it, she freaked out. She was like, 'Oh my God, when I heard these songs, I sat in my chair and I spinned around. I couldn't believe it. I thought you were gonna turn it into that pop stuff. And I was really ready to go there with you. But oh my God, Linda, these songs... you made me feel like I love them again.'

"We just hit it off because we have the same work ethic. There was this funny moment between her and I where we were laying down strings. We were there from eight in the morning to 10 at night and she's like, 'How long you going?' I'm like, 'How long you going?' She's like, 'I'm going as long as you're going.' And I'm like, 'Well, I'm going as long as you're going.' So we literally sat there 'til two in the morning until all the strings were done. We just laughed because neither one of us wanted to say we were tired or call it quits when we were so close to being done.

"I mean, Dolly Parton is fucking awesome," Perry enthuses. "It will always be the most memorable moment in my life."

It required all of Linda Perry's powers of persuasion to convince Christina Aguilera to keep the original demo vocal that made 'Beautiful' such a powerful recording.

Keeping Scores

Following *Dumplin'*, and her soundtrack for Sean Penn's 2020 documentary, *Citizen Penn* (concerning the actor's involvement with relief efforts in Haiti after the country's devastating 2010 earthquake), Linda Perry is now focusing more on scoring, and writing fewer songs.

"It was very disappointing how many artists came in and wanted to sound like somebody else," she admits. "I mean, even asking me for [*copies of*] my own songs. I'm like, 'No, I've already written it. I don't double dip.' I've never written another song that sounds like 'Get The Party Started' or 'Beautiful' ever again. If you want something original, I'm the person you come to. But if you want a soundalike, go somewhere else.

"So I stopped working with people and I started scoring. I'm starting all over, you know what I mean? Like, I'm not Linda Perry, songwriter 'legend', when I walk into trying to get a job as a composer. They don't give a fuck. I'm just a new person that is starting from scratch.

"Scoring is a completely different world," she adds. "It takes a completely different type of patience, and a completely different skill set. And so, I'm learning the craft right now. I have a few things under my belt, and I'm OK with that.

"I'm used to starting over. I've reinvented myself so many times that this is no different [*laughs*]. And I will succeed. And I will be one of the best, y'know. But I have to do the work to get there."

Looking back at her 30-plus years as a professional musician, songwriter and producer, Linda Perry now has a clear perspective on the upsides, and downsides, of the music industry. "I'm really grateful for the incredible opportunities this business has given me," she states. "Even though sometimes it sucks. But it doesn't matter. I wouldn't go anywhere else. I love it. I love how sleazy and fucked up it is, and I love how magical and beautiful it is, as well."

Linda Perry: "I'm someone that you can trust, and I'm always going to be honest. And people come to me all the time for my honesty and ask me, 'Hey, what do you think of this?', and they're prepared for a very raw answer."

Lewitt Audio Pure Tube

Valve Microphone

Lewitt's vocal-oriented Pure Tube promises valve tone without the noise.

NEIL ROGERS

From their headquarters in Vienna, Austria, Lewitt Audio have built up a formidable reputation for innovation in microphone design, culminating in last year's flagship LCT 1040: a mic that offers an extensive selection of sonic options around the theme of blending between valve and FET signal paths (see www.soundonsound.com/reviews/lewitt-lct-1040). The impressive features of the 1040 come at a cost, however, and Lewitt have sensibly borrowed some of the R&D that went into that product to produce a more affordable option for engineers who require a valve mic of similar quality but who don't need all the bells and whistles. The result is the Lewitt Pure Tube, a cardioid-only valve capacitor microphone.

First Impressions

Lewitt offer the Pure Tube as part of two different bundles. The Essential Set comes with a minimal set of accessories, including a basic mic mount and an even more basic case for keeping the mic dust-free. I was sent the Studio Set, which features a robust Peli-style case housing the mic, the compact power supply and a very large shockmount, which also serves as a neat way of attaching the custom pop filter. I like this idea of offering both a minimal set and a more fully featured package, and it is perhaps a better option than cutting corners by supplying lower-quality cases and mounts to all buyers regardless of whether they're needed.

The mic itself looks very similar to the flagship 1040, with a simple, functional form factor that makes no attempt whatsoever to mimic any of the vintage designs we are so accustomed to. I often like to ask clients coming through my studio what they think of a review mic's appearance and, for what it's worth, I recall hearing the words 'fancy' and 'modern' more than once. I won't dwell too much on aesthetics, but the design choice of having the valve visible — housed like a trophy in a miniature cabinet — serves no practical purpose and will perhaps spilt opinion. It can look nice in a moody low-light recording session, thanks to a small LED that creates the impression of the valve glowing, but it could also be described as an unnecessary gimmick. Overall, though, I was very impressed with how solid everything looked and felt in use and this, combined with its impressive 10 year guarantee, means you should feel confident about the reliability of what will be a substantial purchase for many people.

Circuit Training

Lewitt are keen to point out that the "revolutionary" circuit design of the Pure Tube delivers unusually low self-noise for a valve microphone. This is said to be due to the absence of any semiconductors or capacitors in the signal path, and the quoted self-noise figure of just 7dB (A-weighted) is very impressive indeed. By way of comparison, for example, the quoted self-noise of the Neumann M147 is 12dBA, and most other current valve models are noisier still. In fact, only a handful of solid-state mics actually better the Pure Tube's self-noise specification.

Although it's not often a problem in my day-to-day recording work, where I'm tracking bands or material that has a dense backing, microphone self-noise can become audible if you use heavy compression and like to layer up your vocals. The Pure Tube's impressively low noise floor also opens up the option of using it in settings where a valve microphone might not otherwise be considered appropriate, such as sparse acoustic and classical recordings, or even voiceover sessions.

Features

The Pure Tube features a one-inch, gold-spluttered capacitor capsule that is clearly visible through the grille, with a lime-green capsule surround that ties in with the look of the overall package. The all-important valve is a 12AU7/ECC82. As the Pure Tube is a fixed-cardioid mic and has no pad or filter settings, there are no user controls at all, and this is reflected in the compactness of the external power supply. If you're contemplating buying the Studio Set, it's worth mentioning that the included shockmount is pretty imposing! Its size shouldn't cause any practical issues, but it won't suit users who prefer a sleeker, more discreet look, and it would be hard to hide in photos or video. It seems beautifully engineered, however, and getting the mic set up and adjusted was refreshingly quick and intuitive. It also allows you to quickly attach the included pop filter so that it's perfectly positioned for vocal recording.

In Use

My routine for reviewing mics at my studio is fairly well established by now: I make a point of trying out the mic 'blind' at first, in order to be as objective as I can, and will initially use the mic in a few 'low risk' settings such as drum room mics or as an extra vocal option, while I get a sense of what the mic is like. I'll wait until I've developed a feeling for its capabilities and decided I can trust it before throwing it up as a main mic with a paying client.

My first use for the Pure Tube was on a cello recording session, where I positioned it as an alternative alongside a Neumann U87 at a distance of around three feet from the instrument. I was immediately impressed with the balanced, clean recording coming back through the speakers. It offered a subtly different tonal quality to the U87, with a softer midrange that pushed the sound of the bow meeting the strings a bit further back in the mix. Next up was a drum recording session, where the Pure Tube served as a mono room mic positioned about five feet in front of the kit. I'm not sure this is a role the designers had in mind for their mic, but I was happy with the slightly trashy mono room option it provided. Cymbals can be a good test of a mic, and the results here were bright but not overly harsh.

After an equally impressive performance in front on an acoustic guitar, where it had all the detail I was looking for, with a nicely balanced midrange, I felt confident enough putting the Pure Tube up to track a male vocalist (recording a Christmas song in July!), and we were all impressed with how it allowed the singer to get very close to the mic, for an intimate feel, without the proximity effect becoming overbearing. This last point is one of the things I noticed consistently when using the Pure Tube in my studio: it always seemed to provide a compact bottom end, even when positioned close to a singer or instrument.

Lewitt Audio Pure Tube
From £859

PROS
- Well-balanced sound.
- Nicely contained low end.
- Good value.
- Solid build quality and 10-year guarantee.
- Shockmount and pop screen in Studio Set work very well.

CONS
- Styling will perhaps split the room.

SUMMARY

The Pure Tube offers the desirable qualities of a valve microphone in Lewitt's thoughtfully designed, well-built, modern style. By omitting any additional features, they've created a versatile vocal mic that will also work very well for general instrument recording.

This can be useful in many situations, but it didn't perhaps strike me as the best option if I was looking to help fill out a thin-sounding singer.

Summing Up

I managed to use the Pure Tube on a varied selection of vocalists over the review period, in styles including acoustic, indie rock, rap and full-on metal. I could tell I was using a valve mic, but in a nicely understated way: harsher voices were softened slightly, and you could hear some subtle saturation on very loud singers. In fact, I found myself liking less how the Pure Tube handled very loud singers as, unlike my U47-style mics at the studio, it seemed to become a little thin-sounding, and generally it provided less obvious saturation and character than you might get from other valve options. The flip side to this, though, was that it always provided a good sense of clarity, and stayed true to what was in front of it.

Overall, the Lewitt Pure Tube is a great-sounding microphone that seems to offer a nicely judged balance between the sound of a valve mic and ultra-quiet, clean operation. At around the same price, there are also good alternatives based on the design of historic valve mics, but if you want a more modern design that offers a bit more versatility — and is very, very quiet! — the Pure Tube could be a great option for your studio. ■■■

£ Essential Set £859, Studio Set £1199. Prices include VAT.
E info@lewitt-audio.com
W www.lewitt-audio.com

Neural DSP Archetype: Tom Morello

Guitar Rig Modelling Plug-in

No synthesizers were used in the making of this plug-in.

JOHN WALDEN

Over the last few years, Neural DSP have quickly established themselves at the cutting edge of guitar rig simulation, with both software and hardware offerings that deliver impressive results. On the software front, the latest release is Archetype: Tom Morello. Like other titles in the Archetype series, Neural have worked in collaboration with the named artist to create a software simulation of their complete rig. So, if you like the idea of having access to the epic guitar sounds that shaped Rage Against The Machine, Audioslave, or any of the other musical projects that Tom Morello has brought his distinctive guitar style to, then Archetype: Tom Morello is bound to intrigue.

All The Rage

Tom Morello has used pretty much the same rig (guitars, amp, cab and compact collection of pedals) for most of his career. Indeed, as he described in a recent 'rig rundown' video on YouTube, the now well-worn mid-1980s Marshall JCM800 2205 amp that sits at the heart of his setup even has various settings marked by each knob; it's a 'set and forget' that's lasted over 30 years! The amp is faithfully represented here alongside his equally long-standing Peavey 4x12 cab, with each element having its own page in the familiar Archetype UI. There's plenty of flexibility too: the user is given the ability to adjust the amp's controls away from Tom's preferred settings, the cab sim page allows you to use two virtual microphones simultaneously (from a choice of 10), with control over their position relative to the speakers, and you can load other IRs should you wish.

The included pedals are arranged across a number of pages and, in terms of the signal chain, sit in front of the amp, in the amp's effects loop, or as 'post' effects. Tom is well known for his use of the DigiTech Whammy pedal and that's included here, but positioned in front of the amp you also get a virtual Dive Bomb pedal, which offers a further pitch-shifting effect, providing a pitch glide either up or down, with the speed of the glide controlled by the expression pedal. It's a cool effect.

True to how the man himself runs his setup, most of the other effects are placed within the amp's effects loop and arranged across two pages of the GUI. The emulated Whammy and wah are on the first, while a simple delay, a seven-band EQ and a one-knob phaser (presumably emulating Tom's MXR Phase 90) appear on the second. The Whammy and the wah are classic elements of Tom's 'sound design', and the delay does exactly what it says on the tin. Tom generally uses the EQ as a boost (set flat but with the level raised) and this tactic is replicated in some presets. The 'post' effects comprise a further nine-band EQ plus flexible and well-featured delay and reverb. They're perhaps less obvious reflections of specific pedals within Tom's rig (which does include multiple delay pedals), but they certainly bring additional versatility in a recording context.

The pedals that inspire the Dive Bomb, Whammy and wah obviously feature

Neural DSP
Archetype: Tom Morello
€143

PROS
- Sounds epic!
- Authentic virtual recreation of Tom Morello's guitar rig.
- Good support for MIDI expression pedals.

CONS
- Morello's rig doesn't cover as much ground as some others.

SUMMARY
Archetype: Tom Morello is a faithful recreation of the artist's unique guitar rig — if that's the sound you want, this is where it's at!

This is a Tom Morello rig emulation so, yes, you get a Whammy pedal — and the plug-in makes setup for a MIDI expression pedal very easy.

The cab sim section provides plenty of options, including the ability to load IRs, and the presets designed by Morello himself are based upon some iconic tracks.

expression pedals, and if you want to replicate Tom's distinctive playing style you'll need a suitable MIDI expression pedal for real-time control. The plug-in's MIDI Learn system makes this very easy to configure, and once you've done that it's a very satisfying playing experience: if you have some Morello licks to hand then Neural's recreation of his rig will deliver this characteristic element of the sound with ease. The Whammy is particularly impressive, with options for setting the pitch shifting ±24 semitones and a detune mode offering ±100 cents. It's obviously a pedal that requires a little practice, but once you get it down it's a heck of a lot of fun to use!

Shredding In The Name

From interviews, it's clear that Tom embraces the idea of getting inventive with a modest selection of gear that he knows well. His compact set of presets recreate the settings used in some of his more iconic tracks and, sonically, they are spot on: just play like Tom (that's the hard part!) and you'll hear his sound. The more general rhythm guitar preset is also right on the money: overdriven and raw. Indeed — and I mean this as a compliment — it's probably the 'wildest' JCM800 emulation that I've tried. It's great for aggressive classic or modern rock tones, and whether you're doing your best Tom Morello, early EVH, Nuno or Slash impression, this amp model will deliver something suitable.

The plug-in also ships with a larger collection of 'factory' and artist-designed presets, and these span a wider range of sounds than you might expect, with some additional ambient patches making great use of the post-amp delay and reverb options. There are even (shock, horror!) a few clean tones to be found here, and generally the amp model responds nicely to your guitar volume control, so you can clean things up in that way also.

The plug-in also offers the other usual elements of the Neural GUI. This includes options for configuring input and output levels, an effective gate for the input, a tuner, tap and tempo options (when using the software as a standalone application rather than within a DAW), a transpose module and a metronome. You also get Neural's rather wonderful Doubler switch with the associated Spread knob to adjust the time offset between left and right channels. This is a really effective way to create a tightly 'double-tracked' effect for rhythm or riff playing.

The Archetype range of plug-ins are an interesting part of Neural DSP's product line. There is absolutely no doubting the quality of the modelling on offer here — it's great. And for fans of specific artists, the appeal is very obvious: whichever signature artist you pick, you get an accurate representation of their guitar rig plus a number of useful extras built into the Neural GUI. In that context, this latest Archetype edition will undoubtedly appeal to fans of Tom Morello, myself included.

But how far can you stray from the Morello sound? As noted above, what's available from the particular amp/cab pairing on offer here can deliver some truly great rock guitar tones, and if that's the palette you're looking for this plug-in would do great service. But if you're after an amp/cab sim to cover a wider range of sounds, it might not be the best option — being a 'signature' plug-in, it wouldn't be appropriate to cram it full of things Morello doesn't use! Don't worry, though, as the same technology is used by Neural in plenty of other 'rig in a plug-in' products, and some offer multiple amps and cabs, as well as options such as overdrive, distortion or compression pedals that can be placed in front of the amp. For example, for one that spans clean, crunch and high-gain sounds, you could check out Archetype: Nolly or Archetype: Rabea, while for cleaner tones something like Archetype: Cory Wong might be a better starting point. (At some stage, I'd love to see Neural deliver a generic front-end that allows owners of multiple Archetype plug-ins to mix and match components to create customised rigs; I imagine this would prove tremendously popular!)

Sound Judgement

When it comes to sound and the underlying quality of the emulations, Neural DSP already had all the boxes ticked. Archetype: Tom Morello is no exception and, while it will not make you *play* like the man himself, it does provide a remarkably convincing simulation of the rig he uses to create his sonic mayhem. Aggressive, angry, inventive and melodic, it's all here. If you want to rage just like Tom, Archetype: Tom Morello puts all the (virtual) gear you need to create those tones into a convenient plug-in format.

£ €142.80 including VAT.
W www.neuraldsp.com

BOX CLEVER

Analogue Warmth In The Box

Mixing in software needn't mean that the results lack character, colour and warmth.

MANON GRANDJEAN

As a mixer, I like to deliver a big and quite hyped sound, and I am always searching for ways to capture analogue warmth. At the same time, though, I need to mix mainly 'in the box' to allow me to work on multiple projects at a time, and have a fairly quick turnover for mixes. To achieve this, I use a combination of hybrid hardware with saturation, distortion, harmonics and console emulation plug-ins. The keys is to add a little bit of all the elements throughout the mix, rather than relying on one plug-in or stage of processing.

Working mainly in the box means my studio setup is fairly simple. My monitors are Ex Machina Pulsars: this is a relatively new brand of speakers, but I was sold as soon as I heard them. They give me the low end I need without a sub, along with top-end sparkle and detail. The same goes for the McDSP Analog Processing Box system. I tried an APB8 and it integrated perfectly into my workflow and gave me the analogue sound that I was missing. A recent addition as well is the Freqport Freqtube, which is a similar concept. I also have an SSL Fusion mastering processor, and my main audio interface is a Universal Audio Apollo x6.

To illustrate how my hybrid mix approach works, I'm going to refer to the song 'Cardboard Box' by the amazing girl band called FLO. It was produced by MNEK and mixed and mastered by me. My aim for this record was to have sparkly vocals recalling 2000s R&B, combined with a big, round low end.

Manon Grandjean is a multiple MPG Award-winning mix engineer whose credits include Stormzy, Dave, AJ Tracy and many more.

Photo: Emma Houghton

Proper Channels

My sessions are usually organised the same way. All tracks go through their respective busses, which I will detail below, which in turn are all routed to a mix bus/master bus.

Once I have done colour-coding, markers, bus assignments and so on, my first step is usually to put a Sonimus Britson Channel plug-in on all my audio tracks and a Britson Bus plug-in on all my busses. This is a console emulation plug-in which I found gives me some warmth and opens my mixes. Each instance has high- and low-pass filters, which I adjust accordingly depending on the source. I use the Fat mode on both tracks and busses, and Master Loudness on the Britson bus. You can assign multiple tracks to a group, but I don't use this feature: I like to set it and forget it.

Once the console emulation is in place, I start carving each track with FabFilter's Pro-Q 3 equaliser. I also switch on my master bus plug-ins pretty early on — usually as soon as my initial gain staging is done. This is necessary because I find that parts always come to me very hot in level; I usually apply -10dB clip gain so that everything is at the optimum level for the plug-ins to run

FLO's hit 'Cardboard Box' puts a modern twist on the classic R&B girl group sound.

Waves' L3 multiband limiter provides both limiting and frequency rebalancing on the master bus.

The four-band enhancer section in iZotope's Ozone can achieve different results depending on the mode to which each band is set.

properly, and not be too high in level going in to the mix bus. If it's not enough then I clip gain down some more.

The plug-in chain on the mix bus changes slightly from mix to mix, but the chain I used on 'Cardboard Box' is fairly typical. First I use the SIR Standard Clip plug-in to catch any kick or snare peaks, but this is only doing some very light clipping (1 or 2 dB) with the saturation down at 4 percent. Then I use the Waves SSL G master bus compressor to give me some glue, applying about 2 or 3 dB gain reduction at most. I set the attack time to 10 or 30 ms and the release time to 0.1 or 0.3, depending on how tight or open I want the mix to be, I also always stay on 4:1 ratio.

After the SSL, I use the Waves L3 Multimaximizer to impose a 'smile curve' frequency response on the mix. I then add Slate Digital's Virtual Tape Machines plug-in. I like how this brings the mids forward and shapes the low end. After that, I'm using iZotope's Ozone 9 for some multiband compression, overall EQ, enhancement and stereo imaging. On the exciter section, I prefer the 'warm' setting on the low end and low mid bands, but in this case I used 'tape' on the high mids and high end to make the vocals pop. If the track is quite dense in the upper mids, I would switch the high mids and high end setting to 'warm' as well. Following Ozone, I want to bring back some transient impact, which I'm doing with the Slate FGX Virtual Mastering Console. I bypass the compression section as I don't need more compression at that stage.

We are now transitioning into the mastering part of the mix bus. I master a lot of the mixes that I do, and if I'm working on a single song rather than an album or EP, I do the mastering within the same session on the mix bus.

This is where the McDSP Analog Processing Box comes into play, and here I am using the APB L-18 analogue limiter. I love the Color setting, which adds top end to the mix. I limit very lightly with this plug-in — about 1dB gain reduction, with a soft knee and short release time — as the limiters after that will do the heavy lifting. These are the Softube Weiss MM1 and the FabFilter Pro-L 2. The Weiss gives me a bold, loud sound which I always like »

Manon Grandjean's first mix move is usually to place Sonimus' Britson Channel and Britson Bus on every track and bus in the session.

McDSP's L-18 analogue limiter plug-in for the APB system provides a colourful alternative to digital limiters.

Manon Grandjean prefers to use several limiters in series rather than just one.

» on my mixes, and the Pro-L 2 rounds everything off. The 'Allround' limiting style is my favourite, as it seems to bring in some more low end. I much prefer having a few limiters doing the work rather than just one, as each one has a different colour. With all of them in place, the total amount of gain reduction is about 6dB. The choice can be genre dependent: if I find that the master is missing a bit of punch, I would swap the Pro-L 2 setting to 'Punchy', or use a different limiter like Newfangled Audio's Elevate, for example; I often use this on rap tracks as my last limiter.

Vocals

Once I have de-essed, equalised and slightly compressed (usually with Waves' Renaissance Vox) each lead vocal track, they are all sent to a lead vocal bus. On this song, I used XFER's OTT on the lead vocal tracks, as it carves out some of the mids and brings some top-end sparkle. I use the same process on the ad libs and the backing vocals: treating the individual tracks first then sending them to their respective busses, and using the processing on the bus to fit them in the mix.

On the lead vocal bus, I have a set chain which I use a lot. The UAD Tube-Tech CL1B adds some light compression and also shapes the tone, and it's followed by the Waves CLA Vocals plug-in. I really like using the stereo mode on the Pitch fader to give the vocal some width, despite the common belief that vocals should strictly be mono. On the Compress fader, my favourite settings are Push and Spank. I use Spank when the vocal really has to be in your face, but in this case Push was sufficient.

I am always looking for way to bring warmth to everything in the mix, so Soundtoys' Little Radiator is on all my busses, including the vocal bus; it is then followed by the Eiosis Air EQ. The setting I use boosts the 'air' band but with a dip at 10kHz so it is not piercing.

After this I do more light compression with another favourite plug-in for the APB: the Chickenhead compressor. I particularly like the Sauce mode, which is slightly warmer than the standard mode. The final two plug-ins in the chain are Avid's Lo-Fi, with a tiny amount of distortion and saturation to bring the vocals forward in the mix and make sure

The Pitch fader on Waves' CLA Vocals plug-in has a stereo mode that introduces some widening.

A broad air boost is combined with a 10kHz dip to make the lead vocals bright but not harsh.

Another analogue plug-in for the APB system, Chickenhead provides characterful vocal compression in Sauce mode.

Spot effects such as delay throws are created by copying individual words to a separate effect track.

they cut through, and a Waves L2 limiter to control peaks.

Sends & Sensibility

In terms of effects, I have sends from the lead vocal going to a plate reverb from Liquidsonics' Seventh Heaven, a hall setting from Valhalla Vintage Verb and an eighth-note delay from Waves' H-Delay. The levels of these sends are ridden all the way through the song using automation. Obviously, the eighth-note delay is sync'ed to the tempo of the track, but I also like to match the reverb settings to the song tempo too. For example, in this case I set the pre-delay time of the plate reverb to 46.58ms and the decay time to 2.9s; this makes the reverb tail last exactly one bar at 80bpm. This avoids having reverb tails overhang longer than necessary and crowding the mix. I dial these settings in by calculating beat and bar times for the song tempo, then adjust by ear if necessary.

For spot effects such as reverb throws or quarter-note delays, I prefer to set up a separate effect track and simply drag the audio clips that I want to effect onto this track. This is quicker to set up than using automation to achieve the same results.

Lastly, I also have a send going to an aux track with the UA Thermionic Culture Vulture plug-in, which I adjust depending on how much I need it, to bring excitement to the vocals and bring them forward. This is very useful to help vocals cut through dense tracks.

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Molten Motion Meter

Eurorack Module

Here at *SOS*, we pride ourselves on an unbiased approach to reviews, treating all developers with equal respect, big or small, new or established. Many of them we know personally, many we don't. But I've never had something pass across my desk from quite as close to home as the Molten Motion Meter, since it was developed by our very own Robin Vincent; a core contributor in this here modular corner of the magazine, as well as to *Sound On Sound* at large. Molten Modular is the name under which Robin has recorded his foray into Eurorack online, featuring a host of rather excellent videos, a newsletter and now a burgeoning line of brand-new modules. It's well worth checking out his website (https://moltenmodular.com) and accompanying YouTube channel, particularly if you like jazzy shirts.

The Molten Motion Meter is the fruit of a collaboration between Robin and DIY-leaning Eurorack company Befaco. Available either assembled or in kit form, it came about in response to what Robin calls "my desire to be able to visualise modulation... to be able to better connect visually and aurally to what the heck was going on with the modulation and signals in my patch."

So what is the Motion Meter? Well, it can be a lot of things. Its cute 6HP panel presents three knobs, each surrounded by a halo of LEDs. Each of these has its own input, output and accompanying three-way mode switch with options for Audio, CV Attenuation and CV Inversion. With its three outputs able to mix down to Output 3, it can also act as a simple mixer, with some handy three-colour visual metering to boot. I'll also say I was pleased to find that at full tilt in Audio Mode it's actually able to drive sounds a little into what sounded to me like clipping distortion, which you may like or hate, depending on your persuasion. Switch to CV Attenuation and the LEDs around the knob in question turn purple. With nothing patched to that channel's input, it outputs a static 10V at maximum, suiting it well to acting as a remote control for another less reachable or less visually intuitive module during live performance. Even simply using it to open a filter is satisfying, with that accompanying ring of purple LEDs. Patch any CV into that channel's input and the LEDs will meter the motion (geddit?) of the incoming signal, and the knob becomes a useful offset.

Set a channel's switch to CV Inversion and its respective LEDs turn amber, metering anti-clockwise around the knob instead of clockwise, which is a nice touch. This is where the Motion Meter becomes a lot more creative, since it can invert any signal thrown at it and completely change the character of a signal — even a patch — at the flick of a switch. With nothing patched to Outputs 1 and 2, Output 3 will sum their values. This makes for some great modulation possibilities: set one to attenuate and the other to invert, for example, and work the two against one another. I particularly enjoyed routing the main body of a simple patch through the Motion Meter to control every aspect of it. Through Channel 1, a VCO; through Channel 2, CV from a sequencer to that VCO; and through Channel 3, an envelope going to that VCO's wavefolder. This way I could adjust the volume of my oscillator, attenuate its pitch and adjust the amount of wavefolding all from a single panel. Classy!

If there's one criticism I could throw at the Motion Meter, it's that it would benefit hugely from CV inputs for each channel's parameter position. On one or two occasions I found myself wishing I could have the knobs' values move by themselves, or at other times to move faster than one could turn the knob manually. Apart from anything else, this would also essentially render it a fully functional three-channel VCA on top of all its other functions, which would be tremendously useful. That said, I respect the way that the Motion Meter encourages — even demands — hands-on performance, and this is much of its identity at the end of the day. One assumes it needed to go one way or the other: patchable and socket-heavy, or big, beautiful knobs and LEDs. Clearly the latter won the day, and as it is there's not much to complain about.

The Motion Meter is also available in a recently released 1U version, which offers four channels over the original three. This is a good move, with that extra channel opening up a lot more combination opportunities — particularly when it comes to working with stereo signals. Whichever version speaks to you, the Motion Meter is a great example of what I think of as an 'enabler' module, expanding the functionality of all the modules around it and so open-ended as to only be limited by one's own lateral thinking. This functionality would expand exponentially with more than one in your case, which I can totally imagine having since it's so widely applicable, and I'll submit it's just about affordable enough to warrant repeat purchases. Bravo, Robin.

William Stokes

£ £149

W www.befaco.org

Worng Electronics ACRONYM

Eurorack Module

In the biscuit barrel of modular, ACRONYM is a chocolate Hobnob dunked into a chilli mochaccino. It looks like a regular biscuit but has a thickness, a texture that holds it together and an edginess that crumbles with just the right bite. As you dunk it into your Eurorack beverage, it soaks up harmonics, absorbing timbres that flow and dribble. The aftertaste kicks in with heat from the sync and through-zero phase modulation. And just when you were expecting a soggy conclusion, the subs lean in for a surprising chewiness. I should have my coffee break here more often.

ACRONYM is an acronym for Analogue CoRe Oscillator New tYpe: Morphing, which feels like Worng are trying a little too hard to find an interesting name for a module that can speak for itself. Everything about this module draws your attention. The strangely scribbled gold text on a dark chocolate front panel is very eye-catching, as are the fierce red LEDs illuminating mystical symbols. The pulsing inverted triangle is just off-centre enough to be troubling, and the whole thing feels edgy. All of that vibe pours through to the waveforms, which are definitely angry about something.

At the bottom of the module, you get five outputs. The first two are the sine and triangle outputs, which are perfectly sweet. The following two are the -1 and -2 sub-oscillator outputs, which are not entirely square, allowing them to be badly behaved in self-patched audio-rate modulation. The last output is the wavemorphing out, which is where the fun begins. The analogue wavemorphing pushes the core through some harmonically rich transformations. Starting from a humble triangle wave, it flexes into a square and smoothly collapses into a sawtooth before folding in on itself for some nicely pure organ-type tones. The transitions are smooth and get deliciously fluid towards the wavefolding. A bipolar CV input gives you many ways to play into those transitions. The movement is fabulous, from short envelope rasps to long-flowing LFOs. If you need a bit more beef, you can mix in the sub-oscillator to broaden the frequency content.

Talking of modulation, the ACRONYM opts for through-zero phase modulation rather than the more common FM. Through-zero has been a revelation for people like me who struggle to get anything tuneful out of audio-rate modulation. A little bit of movement into the PM input gives it a chewy detuned vibe, whereas driving another oscillator into it generates a controllable chaos of FM-flavoured clangs and tears. Once you hit those ratios and pull the modulator in line with the ACRONYM you can hit upon some wonderful tones. To top it off, patch the same oscillator into the sync input, and you have a tremendously characterful output, dripping with edges and animated movement.

The ACRONYM offers a huge amount of tonal variation that feels much bigger than an analogue oscillator should. It looks fabulous, it's spacious and easy to work with and drags your attention away from the filter you're inevitably running it through. The phase modulation and wavemorphing are so enjoyable that I didn't notice the lack of PWM or miss a dedicated FM input. It's a strong contender for my new favourite oscillator.

Robin Vincent

£ £395

W www.worngelectronics.com

Modular Profile: Arushi Jain/Modular Princess

Modular synthesist, singer, creative technologist, engineer, composer, producer, radio host, coder; the list of Arushi Jain's, aka Modular Princess', abilities goes on. Based between New York and New Delhi, Jain has toured the world over with a critically acclaimed brand of electronic composition, infused with the influence of Hindustani classical music. Jain is also a talented instrument designer in her own right, not least demonstrated by ektara, a tool enabling her to sequence her synthesizer by singing a melody.

On her entry into modular

I was introduced to sound synthesis at the Center for Computer Research in Music and Acoustics at Stanford University, where I was an undergraduate studying Computer Science. The community I found there was very experimental and progressive, and some of my close friends had modular equipment which I was always incredibly intrigued by. I was too shy to ask about it though, and so it wasn't until 2017 when I was back in Delhi, India when I was first able to spend some time with one. It was love at first sight, and a month later I was back in San Francisco and had acquired my first case; a beautiful custom wood Bastl that is still my favourite case to date.

On her go-to modules

Right now, my staples are [*Mannequins*] Mangrove, [*Make Noise*] Maths, [*Make Noise*] Morphogene, [*Expert Sleepers*] Disting EX, [*Mannequins*] 3 Sisters, [*ALM/Busy Circuits*] Pamela's Pro Workout... and a whole lot of Expert Sleepers! But to be honest, this changes depending upon the album I'm writing. For sequencing, I use »

» a combination of my computer, keys, [*Frap Tools*] USTA, and my voice.

On combining Indian classical music with modular synthesis

I'm not an Indian classical musician, I'm a modular synthesist. I have a deep interest in Hindustani classical music — the traditional music of North India — and in fact it's the only musical language I speak. I studied this music primarily as a vocalist while I grew up in Delhi, and I will continue to take classes forever. I'm still a beginner; this music is so vast there is so much to learn, and to be quite frank with you, it is like holding up a mirror: the more I learn about Indian classical, the more I learn about myself.

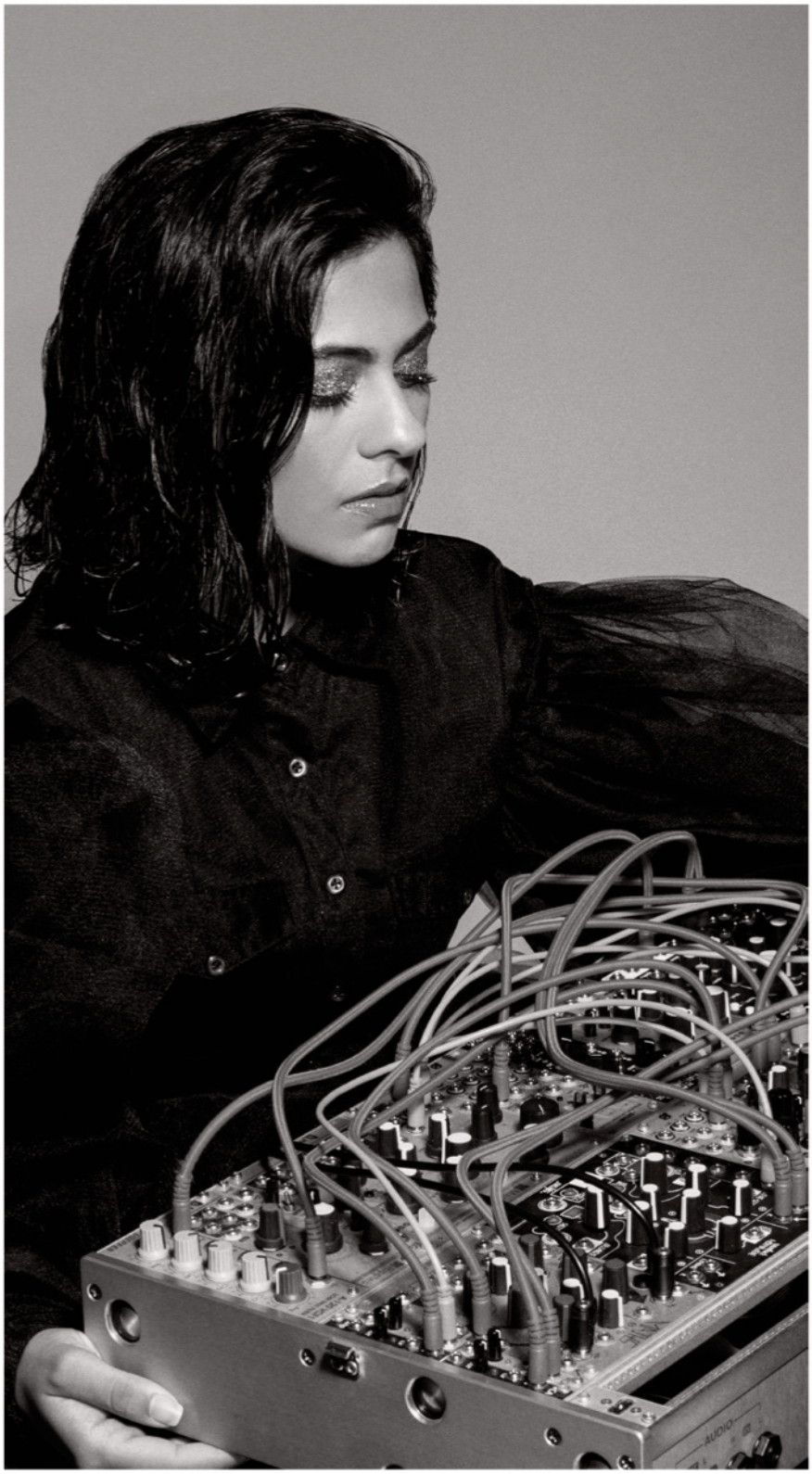
I am interested in experimenting with the sonic identity that Indian classical has had for centuries, and stretch it until it creates an entirely new form. I am not interested in replicating Indian classical music on the synthesizer. There are new planets to be discovered, and I prefer to remain free to walk wherever I please.

On building ektara

My synth is my bandmate, and I wanted it to have the ability to respond to me, copy me, mimic me, riff off me. This was the core inspiration behind ektara. It enables me to sing a melody into a microphone and directly sequence my synthesizer. Using a pitch-detection algorithm in offline mode, my voice is converted into MIDI notes and pitch-bend data to enable microtonality in synthesizers. Using Ableton and an Expert Sleepers FH2 MIDI-to-CV converter, I can sequence my modular synth with CV using any DC-coupled interface. I'm sure there are tools today that you can buy on the market to do exactly this, but I personally really enjoyed building software and like to make tools for myself, to improve the quality of my life, and art.

On the culture of modular

I think the modular synth world is so unique, and affords the synthesist room for so much personal expression. I love the fact that you build your own case, which makes everyone's sound a unique match to their individual flavour. I find a lot of similarities between me and other synthesists. People who like to hack on machines are often quite entrepreneurial and have a logical forward brain. I also find that people are generally quite nice, and excited to problem solve with you.



People say the modular community is too much about the gear and the technology, and not about the music and art. This might be true, but it doesn't bother me. Any instrumentalist who is an expert at what they do intimately knows the architecture of the instrument that they are playing, not just how to play it. I think it's important to be curious about gear if you're working with modular synthesizers, because if you're not, then how will you learn? *William Stokes*

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TASCAM

WIN! Worth £2099
Focusrite ISA 428 MkII

Few preamp designs can boast a history as illustrious as that of the Focusrite Input Signal Amplifier, or ISA. The original ISA 110 preamp module was designed by none other than Rupert Neve, as part of a console specially commissioned by Sir George Martin for his then-new venture, AIR Montserrat, in 1985. Shortly thereafter it found its way into the coveted Focusrite Forte console, and has since been re-released in a variety of formats.

For this month's exclusive *SOS* competition, we've teamed up with Focusrite to offer you the chance to win one of the ISA preamp's latest incarnations: the ISA 428 MkII. Boasting no fewer than four ISA mic amps in a 2U rack, the 428 MkII uses exactly the same circuitry and Lundahl input transformers that gave the original its sound. But it also incorporates a number of conveniences that make it ideal for both studio and location recording.

Each input has its own front-panel DI socket, so you can capture instruments without disconnecting your mics. Every channel also features a high-pass filter, variable from 16 to 420 Hz, while switchable input impedances give you a range of tonal options and ensure optimum performance with any mic, from vintage ribbons to transformerless capacitors. Stepped gain controls with variable trim make it easy to set exactly the right amount of gain.

To enter, please visit:
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Round the back, in addition to the expected mic inputs and line outs, you'll find additional line inputs, plus balanced insert sends and returns. There's also a slot for an optional digital expansion card, which endows the ISA 428 MkII with ADAT, Dante and AES3 connectivity.

Finally, as you'd expect from a Rupert Neve design, the specs are examplary, with up to 80dB of gain, a huge signal-to-noise ratio of 123dB, and a frequency response that extends all the way up to 94kHz for the mic inputs.

To be in with a chance of winning this fantastic preamp, simply follow the URL shown, and answer the questions there, by Friday 6th October 2023. Good luck!

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Steinberg Dorico 5

Notation Software

Steinberg's Dorico inches ever closer to full DAW status...

MARK NOWAKOWSKI

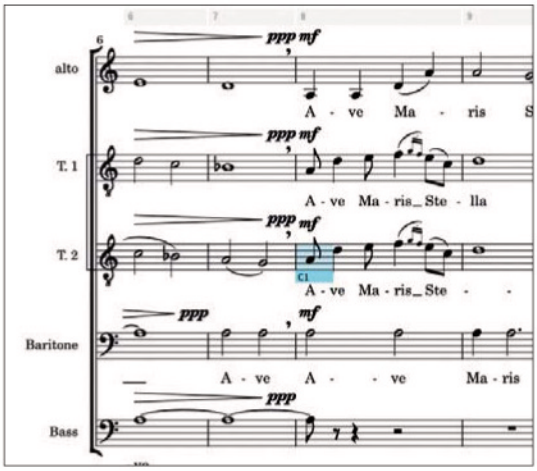
The evolution of Dorico from a notation application contender with a storied development cast and some real potential to the innovation-driving rising star it has now become has been well-documented in the pages of *SOS*, including the reviews of Version 3 in the February 2020 issue and Version 4 in June of 2022. And while program updates seem to be coming faster than ever these days, it is still fair to ask: is it really time for another Dorico upgrade? Given that music engraving is an ancient art whose relationship to computers started at the University of Illinois over 60 years ago, just what else is left to do in this realm? If Dorico 5 is any indicator, it turns out that much is still possible in this area. Dorico 4 established the franchise as a deeply functional tool for music engraving, and now Dorico 5 — while still adding some significant new engraving upgrades — takes a major leap forward as a production workstation. In the final analysis, it is easily a sufficiently significant update to warrant the new number and price tag. Let's dive into some of the highlights.

What's New

One of the remarkably useful additions in Dorico 5 is the inclusion of Stage and Space Templates. With this feature, the individual staves or instruments in the score can be spaced on a virtual sound stage, including the mimicking of stage depth as well. Users may navigate to the mixer toolbar and click on the Live Stage button to open the virtual stage. The instruments can then be dragged forward, backwards, and left to right on the stage. There is also the ability to click and drag to select a larger group of instruments, and then proportionally expand or contract the space in an existing balance with a single click, helping with quick edits and mix comparisons. Predictably, left to right shifts modify the panning in the mix console, while an increase in stage depth increases the send level to the chosen reverb to mimic distance and depth.

The settings related to Live Stage are accessed via the Live Space button, wherein the limit of panning may be set, as well as the level of the chosen reverb. Users may also pick two different reverbs for their near and far sends, respectively. While this new feature is clearly directed towards composers less comfortable with the audio mixing process, power users should not scoff at the usefulness of this feature, as it presents a far more intuitive approach to mixing with an ear towards creating an optimum balance on a realistic sound stage. Even as a starting point for achieving a realistic balance when using virtual instruments (even those recorded in situ) it is superb. For users who merely use the software for notation and playback as a composition aid, the sound stage is equally useful: having parts spaced realistically on a sound stage ensures a cleaner ensemble sound, and the ability to hear much more compositional detail than when instruments are simply all jammed together in the centre of the mix. While using Dorico 5 to compose my annual polyphonic work for the vocal quintet His Majesty's Men, I noticed that having nice spacing certainly aided my compositional experience.

Also new, Pitch Contour Emphasis in playback ensures immediately more musical playback straight out of the box. Simply put, the software uses new algorithms to analyse musical lines and the balance of phrases to attempt a more human and realistic playback. The results are immediately audible, but users need to toggle the feature on when importing scores into the new version in order to hear this difference.



MIDI Triggers allow users to send additional info to VSTs without cluttering scores.

Keeping with the theme of upgrading Dorico's production capabilities, virtual instrument integration has been further enhanced with the inclusion of MIDI Triggers. No longer will the nightmare of scores littered with uncounted ledger lines in order to trigger keyswitches trouble those who have integrated virtual instruments into their notation software. Whether you are looking to drop the aforementioned keyswitch into your score, or just trigger appropriate playback in a pattern-based chordal instrument such as a strummed guitar or patterned drum VST, MIDI Triggers provide an elegant (and, thankfully, easily hidden) layer to accomplish this. Just bring up the Shift+0 pop-over, type in your desired note, and the trigger is added to a hideable layer in your score.

Expanding its already impressive arsenal of features, Dorico 5 welcomes the arrival of Groove Agent, Steinberg's acclaimed drum plug-in. From pulsating grooves to intricate drum patterns, jazz bands, rock rhythm sections and contemporary pop, Groove Agent offers a wealth of high-quality sounds and patterns that will undoubtedly inspire and elevate your music. It's a powerful addition which also opens a new door into Dorico's unabashed evolution into a fully capable production environment.

Live Editing

Daniel Spreadbury and the creators of Dorico have long held to the position that they want to make existing music notation entries as difficult to accidentally edit (and destroy) as possible, which has informed their note input framework. Yet ever mindful of user requests, the newly added Live Editing now allows users to make real-time adjustments without compromising that original vision of safety. Users can drag individual notes or longer passage left or right in the score without modifying the rhythmic placement of later or earlier passages, or drag to transpose up or down diatonically. When such an edit is being made, an orange box indicates the final placement of the selection, allowing for precise movements. The option to copy sections via Alt-click-drag has also been added. A particularly nice touch is the option to click-drag a copy up or down above notes, allowing the building of parallel harmonies. The safety feature to prevent unwanted edits is maintained by only allowing one direction (up/down or left/right) instead of universal movement around the score. Conveniently, the feature also works with chord mode and insert mode.

But Wait, There's More

A host of other smaller but significant updates and new features abound. Scrub Playback allows the in-score unidirectional drag/scrub playback of a score, especially useful in analysing a chord progression or inner voice work while composing. The import and export features related to music XML has been improved. A new Instrument Editor allows for the creation of instrument types not included in Dorico's existing library of 600 instruments. There are also some additional music entry options, especially regarding group editing. Users can now highlight multiple instruments in a score and enter indications with a single command as opposed to having to enter each item separately. There is also some intelligent assistance built in to this functionality, as (for instance) users selecting a section of score with staggered entrances and then applying a group dynamic will see this dynamic placed at the musical entrance as opposed to evenly on a blank staff. It's little changes like this which ultimately save hours in the final engraving stage. Dorico 5 also includes eight new music fonts, easily switchable from the Library menu.

Conclusion

Dorico 5 builds upon the solid foundation laid by its predecessors and introduces a wealth of new features that make it an even more enticing proposition for composers, arrangers and music engravers. From the revolutionary Stage and Space Templates to the nuanced Pitch Contour Emphasis and the seamless integration of MIDI triggers, the software offers an exceptional blend of creativity and functionality. With the addition of Groove Agent SE and the live editing capabilities, Dorico solidifies its position as a top-tier notation software. And with version 5, Dorico moves us much closer to the dream of having a fully production-capable notation software. Highly recommended. ■■■

£ Dorico Pro 5 £497; upgrades from £85. Dorico Elements £85; upgrades from £25. Prices include VAT.
W www.steinberg.net

Steinberg Dorico 5
£497

PROS
- A wealth of new features.
- Easily the most production-friendly notation software on the market.
- Many new options added to an already impressive array of time-saving features.
- Stage and Space Templates are power tools for both accurate playback and later production.

CONS
- A bit pricey for new users, though a free trial, cheaper Elements and free SE version allow a gentle entry.

SUMMARY
Dorico's latest update makes it the most production-capable notation software on the market. Stage and Space Templates are a great GUI shortcut to better mixes, playback is more realistic, and MIDI Triggers de-clutter your score and create new musical possibilities. Multiple item creation, live editing and new fonts upgrade the engraving side of things.

Elements

Reacting to recently collected market data, Dorico's latest 'light' version, called Elements, sees major upgrades that should make it more appealing across the beginner, student and amateur demographics. Elements includes unlimited ensemble sizes, expanded engrave mode with deeper spacing controls, customisable playback options and more text and music fonts. It is enough for me to make it the centre of a youth composition seminar I am teaching this summer. And for those looking for a robust free tool, Dorico SE remains a strong option.

TalkBack

Mörmaid

WILLIAM STOKES

Norwegian artist and producer Live Sollid began her musical journey in the world of jazz, but is now exploring the outer reaches of progressive electronica and experimental pop, with a palette of otherworldly vocal manipulations, lush soundscapes and deft experiments with beats in Ableton. For Mörmaid, it's all part of the same, vibrant ecosystem. "Jazz vocals and improvised music are the base of what I do," she explains. "It's always been a part of it, particularly when I talk about production."

At the moment I can't stop listening to

This maybe isn't going to be the most exciting answer: I recently had this long period where I actually didn't listen to that much music. Instead I've listened mostly to podcasts! The stuff that I keep going back to includes people like Jon Hopkins, also Kelly Lee Owens. Her album *Inner Song* is one of my favourite albums. I've got this Spotify playlist where I gather all of my favourite electronic music. Also, lately, I've been listening to this band called Giant Swan. I think the two first tracks on their self-titled album are actually my favourite tracks: '55 Year Old Daughter' and 'Pandaemonium'. But I also love their latest EP, *Fantasy Food*.

The project I'm most proud of

I want to say my first EP, which I did all by myself. It was during the first lockdown that I finished it. I recorded, produced and mixed it on my own. And I felt really good about having done all of that myself. I felt like I got the mixes to a really good place. In the beginning, it was really hard to have any motivation at all, to do anything. So I just got into this mindset of, 'OK, fuck it, I'm gonna do something. I have the songs.' I just felt, 'Wow, my ears can do this! I can mix, and it sounds good.' I was referencing it next to other stuff and I was really happy with it. So that's something I felt really, really good about.

Also on my recent EP, the track I'm most happy about is a song called 'Punish'. Because it's a really simple idea, but it works really well, I think! The song title actually just came from the [*Soundtoys*] Decapitator plug-in, which has a 'Punish' button! I was like, 'Punish! Oh, OK, I'm gonna play with that word a bit, and then stick it in a lyric,' and it is really simple! And of course, all of the stems basically have the Decapitator on them. I use it all the time!

The first thing I look for in a studio

Oh, good question. I think, if there is a synth rack, I like looking at what's there. I'm really interested in synths and I want to learn more about them, because I feel like I'm just like touching the surface. So I think that's the first thing. I'm like, 'Ooh, what's this? What's that? What does this thing do?' I have a [*Dave Smith Instruments*] Prophet-12 and a [*Teenage Engineering*] OP-1. Those are my two. I feel like I still have a lot to learn, at least about the Prophet because it's such a huge beast. The OP-1 is really fun. I feel like it really has a character; you can just hear if someone has used one, even though it has a lot of different sounds, it's got this kind of wacky character.

The person I would consider my mentor

Well, my mum is definitely my hero! But actually, I'd say two of my singing teachers have both been really important for me. The first one I had in college, when I went to do music there. Her name is Nina Lundberg. She was my singing teacher for the three years I went to college. And I hadn't really had any singing training before I started with her, so my voice was very unpolished! But she she was so super helpful. Helping me, like, discover my voice and what I could do with it. It really opened up so much. And my voice changed like a lot during those three years. Suddenly I could work across a really big range. Also she made me feel super safe and was a very caring person. Just so generous. I don't know where I'd be without her, when it comes to singing.

My other singing teacher was very different. I had her when I did my undergrad in jazz vocals. She was the one who introduced me to doing very experimental stuff, or extended vocal techniques. Her name is Sidsel Endresen and she's a really well-known, highly respected musician here in Norway. She's considered a pioneer when it comes to experimental vocals — and also lyrics.

My go-to reference track or album

I really love SOPHIE. It's so clean but also so crazy. The way it's mixed, it's often just so satisfying to listen to, it's kind of almost ASMR, you know? Quite bright, but also with this really nice low end to it. So I often use her music as a reference. I would say 'Faceshopping' is one of my favourite tracks, also 'VYZEE'. It's weird, it's very dance-y. It has this pitched-up voice, kind of speaking or rapping, almost. Which is not SOPHIE herself. It's almost a bit uncomfortable, but it's also very comfortable! You get kind of confused listening to it, but I love it. All the elements have their place. Nothing is unnecessary, you know. It's just really well balanced and well placed in the soundscape.

My top tip for a successful session

When I work by myself, I feel like it's been a successful session if I've discovered a new way to do something, or a new way to use an effect with my vocals. Maybe just something that I haven't heard before. That's where I feel I get that kick. So, that's all what I think constitutes a good session. And it doesn't even have to be something that sounds catchy or anything. It could

Photo: Guro Sommer

just be an interesting sound design thing or something.

The studio session I wish I'd witnessed

I would have loved to be there when Burial did the song 'Archangel'. I think that's so amazing, so beautiful. It would be so cool to be... In Norway we say a 'fly on the wall'. I don't know if he just did it himself. I think it was just him, right? I guess that's why I would have loved to be there! The vocals in it are chopped up in this really cool way. And it is quite mysterious. You don't feel like you're getting very close to whoever wrote it, but at the same time it feels very emotional. It's as if it's got this veil. I'm actually working on a track now, where I've tried to make a garage beat like that. I'm trying to find a cool way to sort of put a little nod to Burial in there.

The producer I'd most like to work with

I would absolutely love to work with Jon Hopkins. I'm a huge fan. There's a lot of detail, but also it's very simple. I went to see him play here in Oslo last autumn, and he was mixing up these very dance-y electronic tracks that he has with his piano pieces and with strings. It was super cool, that contrast. It worked so well. His album with King Creosote is also one of the best albums I've ever heard. The first time I listened to it, I was just amazed! I was crying, feeling everything! That's something I would love to do myself, actually: work with someone whose sound is so different to mine and to make it work so well.

The part of music creation I enjoy the most

What I like the most is to just improvise with my voice. All songwriting is a sort of improvising, but in the starting phase it's about using the stuff that I've learned, with a very open space. I'm trying to not be too focused on genre or anything; instead, just seeing what comes out. Of course, it does often end up in this electronic landscape, because those are the tools that I use and that's also what I really like to use. But yeah, I just really love improvising, and especially with my voice. There's so much you can do with the voice apart from just singing a melody, and there's so much textural stuff you can do, so many sounds like that you can create, even without any effects. So I really like to come up with ideas that way, just using the voice.

When I did my jazz education, for the people in my class who played other instruments, there was a focus on learning to sing something before you learned to play it. Of course, I'm a singer, so I would do things with my voice anyway, but if we were learning a chord progression or a bass line or something, we would always sing it first and then move on to the instrument. So there's something about using the voice as this basis, because it's so close: it's in you. When you learn to sing it first, it really stays in your body, you know, and it really becomes a part of you. I think that comes out when you move onto an external instrument as well. It's very primal, in a way. And it really does something to how you perform. I love using my voice as a tool.

The advice I'd give myself of 10 years ago

Ten years ago, I was 20... being in this jazz community, people felt a lot of pressure to know about all of these obscure things, to know all the names of all the jazz musicians and so on. There was pressure to be able to play very advanced stuff and to be very technical. It obviously wasn't like that all the time. But it was definitely a part of it, because there is a lot of like, shredding in jazz, you know? So I think I would maybe say, you don't have to be be able to nail those crazy jazz licks. And you don't have to pretend to like all the jazz musicians. You can just like the music that you like, and do whatever you're interested in. I remember thinking, 'OK, I'm a vocalist, but I'm not going to be one of those singers who just does melody and lyrics, I'm going to be an instrumentalist vocalist.' And that's something that part of me actually thought was very cool, but I also think I was trying to prove something. Trying to prove that I was not just a singing lady, you know. You don't have to prove yourself more than, for example, the male jazz musicians. Just do what you're passionate about, and it's gonna be great because you're passionate about it!

VOICE Craft

The Art Of Vocal Production

The technical dimension to vocal production is important, but the human dimension is even more so.

RAMERA ABRAHAM

The vocal recording area doesn't only need to sound good: it also needs to feel comfortable for the performer.

Whenever I get asked about vocal production and my process, I am often met with questions about how I use compression, my "favourite microphone", and various other technical aspects. I suspect that producers and aspiring engineers are asking these questions because they describe the elements of recording that can be reduced to a straightforward choice, or problems that can be 'solved' by buying something. However, one of the most crucial parts of vocal production — and any recording session for that matter — is reading the room, then navigating the unspoken boundaries that you identify in order to build trust with the artist. This ability to read the room involves picking up on non-verbal cues, knowing when to contribute to conversation, and being adept at responding to people's micro-reactions, probably before anyone in your session even recognises they are reacting.

Every vocal production session is about etiquette, creating connections, and maintaining a safe space for an artist to fully immerse themself in their track. These concepts, unfortunately, are rarely taught. These habits are learned. At entry level, we are used to being the observers in sessions, always wanting to see what pieces of gear our favourite engineers are using, what their settings are, and how they address the room. But once we are part of someone's process and are responsible for capturing the sound of their record, we are no longer observers. We become preservationists.

In Advance

There is a lot of preparation involved before an artist sets foot in the vocal booth. Vocal arrangement is a large part of my vocal production process, and is usually done remotely. The instrumental two-track and a demo vocal are all I need from the artist to create my arrangement. Although I use Pro Tools to record and mix, anything arrangement- or writing-related goes straight into Logic Pro X. My recording template in Logic is a template for my own vocal, which I lay down as guide tracks for the arrangement. It is difficult to guess what an artist's vocal range is without having worked with them for a longer period of time, so I don't try to tailor the arrangement around that. My primary focus is on filling in gaps in the track, between certain sections, and emphasising specific lyrics in a way that serves the song best. After doing a rough mix of my arrangement into the original demo, I share it with the artist to get their thoughts. If the artist isn't passionate about the arrangement, we build it up

together until it's at a point where we are both happy. I'm usually only swapping out a few ad libs here and there, because while making an arrangement I'm drawing inspiration from the artist's influences, as well as my own — all of which I have communicated prior to opening up the DAW-ing board.

This whole process is really about killing two birds with one stone. One, we sketch out at least 90 percent of the final arrangement, which gives us a broad picture of how the track is going to sound before we've recorded any vocals. This way we get a semblance of vocal structure, and can go into the recording session with a goal in mind. Two, using my own vocals on guide tracks is a way of levelling with my vocalist. They might be apprehensive about singing in front of a new producer, for example, and this is a way to show them that even though I'm not the greatest singer, I don't mind overcoming that insecurity to add value to the track.

On The Day

My vocal template in Pro Tools is simple. I'm using a few auxes for basic effects: reverb, delay, parallel compression and width. You will likely see a stock EQ and de-esser, and a couple of different compressors, initially made inactive on my vocal bus. I will have enough of an idea of what the artist sounds like from their demo and other released tracks to create a starting point with EQ and compression, but will make tweaks during our warm-up. However, more importantly, I have to ensure that the artist has everything they need in the booth to stay focused and in the zone.

I print off lyric sheets in the morning and place them on a music stand, along with coloured pens and highlighters so they can make notes on the sheet as we record and as I give them feedback. I follow the same process on the other side of the glass. Artists will sometimes vocalise their preference to read off their phones, but this is too distracting, and I, too, avoid looking at my phone when we are recording. Texts and social media notifications will get in the way, and I think the school-like feeling of note-taking activates a kind of hyperfocus for the artist. This, plus a few fairy lights, candles, dim control-room lighting, and plenty of water and snacks, helps set the vibe for the day.

Some artists have a preference to record in the control room with me, simply to feel like they are in someone's living room where the setting is more intimate, allowing them to feel more at home. Other artists prefer being in a larger space such as a studio live room, feeling like they're performing on stage. Anything that will make the artist feel comfortable takes priority. The presence of label representatives and A&Rs can add extra pressure to deliver exactly what is asked for by the label, while balancing an artist's own creative pace and ideas. My preference is to have just me and the artist in the room. Either way, before anyone else has arrived, I will spend some time in the vocal booth myself, making sure that the headphone mix is well balanced. I also listen to the same headphone mix as the artist for most of the day.

> "I can't seem to multitask with anything outside the studio, but it's important for me to master things like comping on the go and gain staging while simultaneously giving performance directions."

For me, speed and the ability to multitask are crucial. I can't seem to multitask with anything outside the studio, but it's important for me to master things like comping on the go and gain staging while simultaneously giving performance directions. While taking notes on my lyric sheet, I use shorthand cues that help me remember which parts need to be re-sung, which bits need to be more dynamic, where a better place to breathe would be, and so on. Using my own, made-up symbols also means that if the artist ever catches a glimpse of my sheet, they won't know what's on it!

It's Not What You Say, It's How You Say It

We have to be cautiously optimistic when it comes to giving feedback on vocal performance to an artist. Throughout the session, I give the same amount of positive feedback as constructive. Also, I never address the artist in the second person about correcting a phrase. For example, instead of saying "Can you please do X?" or "You were drifting off pitch," I'll use "we" for nearly every piece of feedback: "We're slightly under the pitch here," or "I think we can punctuate that a bit more!" When the focus is on emotive delivery and conveying the right mood through their vocals, I want the singer to feel like we're in it together, as much as possible.

The most important thing to remember when working with artists is that whereas musicians play their instruments, a vocalist is their instrument. Things like tonal quality, expression and character can't be fixed the same way as a trumpet player adjusts their tubing or perfects their embouchure. Confidence — or lack thereof — is the biggest factor that affects vocal performance. If I have a slightest hint of negative body language, or phrase something callously, it could impact the artist's performance. If they don't feel like they're being listened to or respected in their recording environment, their vocal won't reach its potential that day, and the common goal is lost. All my session preparation, from vocal arrangement to sorting out a headphone mix, is concentrated on gaining the artist's trust and earning that respect. Once you have that, you're flying.

Ramera Abraham is a vocal producer and engineer. She was voted Vocal Producer Of The Year at the 2023 Music Producers Guild Awards.

After The Fact

There is a lot of work that happens on my part after the recording session. For the purposes of quality control and ensuring that what the artist heard in the session is only going to be improved, not changed entirely, I take on all vocal editing, tuning and processing myself. These elements are equally important in vocal production as the recording session itself. I always tackle the lead vocal first, so that there is a master vocal that I can use as the reference for tuning and timing.

A popular question people have is about which element takes precedence when recording: pitch or performance. While the obvious goal is to elicit both from the artist, I will always choose performance over pitch. Pitch can be tweaked, whereas once recorded, performance cannot.

MATT HOUGHTON

Yamaha AG08

Podcast & Live Streaming Mixer

Your computer sees the AG08 as four separate USB audio interfaces, so it can be addressed by multiple apps at the same time...

In recent years the market for podcasting and streaming hardware has blossomed and the latest arrivals to the party are Yamaha. Given that they and their subsidiary Steinberg have been in the mixer, audio interface and digital audio game for decades their presence is hardly surprising, but their AG08 deliberately treads a slightly different path from most competitors: its hardware controls are simpler, more akin to those of an analogue mixer, and there's no onboard recording facility — the AG08 is very much intended for use with a computer or iPad. While this means you can do a little less using the hardware alone than with, say, a RodeCaster Pro, Zoom PodTrak or Tascam Mixcast, many will find the controls simpler and more intuitive, and when you wish to dive deeper there's the free AG08 Controller app for that. Crucially, when hooked up to a Mac or Windows computer or iOS device over USB, the AG08 presents as several separate devices, allowing it to function as a conventional 24-bit, 48kHz multi-channel (8-in/2-out) audio interface, and as multiple virtual interfaces for routing audio from different apps out to the hardware, before passing a mix back to your computer or out to another device for streaming or recording.

Hardware

Available in a black or white finish, the AG08 has a fairly compact footprint, with angled side panels rising to support a wider control panel on the top. Weighing 2.2kg, it's portable but also has the heft to stay put on its rubber feet, even with many cables attached. In the box are the AG08, a universal 12V wall-wart PSU, a 1.5m USB-C to USB-C cable (the AG08 can be bus-powered but the PSU is required for use with an iPad; there's no Android USB support), a multilingual safety guide, a Precautions & Legal Information leaflet... and a curiously minimal introduction to the AG08 that directs you to the online product

Yamaha AG08
£738

PROS
- Multiple USB drivers mean several apps can access the device.
- Simple, intuitive user interface...
- ...but the app offers greater control.
- Decent DSP effects/processors.
- Analogue side of things sounds good.

CONS
- Not as feature-rich as some competition.

SUMMARY
This user-friendly mixer is unusual in appearing to the computer as several separate devices — while it can put in a decent performance for music production and podcast recording, it excels when it comes to live streaming.

page and user manual. Leaflets also provide download codes for some capable software: Steinberg's Cubase AI and WaveLab Cast for Mac/Windows (there's a free update to the new v2 for v1 users, by the way), and Cubasis LE for iPad.

The first two of eight rear-panel inputs are mono, with XLR-jack combi connectors accepting mic or line sources, and the centre jack for channel 2 doubling up as a high-impedance (1MΩ) instrument input. The remaining analogue inputs are for three stereo channels: 3+4 are line level on dual quarter-inch jacks; 5+6 have both RCA phono connectors and a stereo mini-jack, the latter taking priority; and 7+8, which is mini-jack only but it's a TRRS type that serves both as an input and an analogue aux output. This last socket is the only means of connecting an Android phone, and on the subject of phones it's worth noting that there's no Bluetooth connectivity, a feature found on many competing devices and whose omission won't bother everyone but some will miss. There are also two stereo analogue outputs: mix out on dual quarter-inch TRS jacks, and monitor out, presented on both XLR and TRS sockets. Completing the rear-panel connectivity are the 12V power inlet and USB-C socket (USB 2), with a switch to select the power source, and an assignable footswitch jack.

On the top panel, each analogue input has its own channel with a fader and mute switch, and the three stereo channels can alternatively receive stereo USB feeds from a computer/iPad; a switch toggles between the analogue line and USB sources. These channels also have a slide switch to route them to the Streaming bus, of which more later, and channels 3+4 have a high/low gain switch which, effectively, is a pad to accommodate hot signals.

All input channels have a signal-present/peak LED, which lights red 3dB below clipping, but channels 1 and 2 are more fully featured. These have a traditional analogue mic preamp gain knob at the top, along with switches for +48V phantom power (with status LED), a 26dB pad and, for channel 2, a line-level/instrument input switch. They also have access to DSP processors and effects. A backlit button engages an EQ plus compressor preset, whose settings can be tweaked using the app. A similar FX button sends the signal to a reverb/delay effect, set using a push-turn encoder that allows you to switch between reverb, delay or both; again, this is tweakable using the app. Each of these two channels also has its own dedicated effect. For channel 1, that's a Voice Changer, which has hardware pitch (±1 octave) and formant shift controls but further effects can be added using the software (more on that below). Channel 2 has a button to engage an amp sim: a preset, but the settings by can be adjusted in the app. Finally, four buttons store and load channel 1 presets, so you can quickly recall any detailed settings you make using the app. (Obviously, this only recalls the digital settings — not, for example, the preamp gain or phantom status.)

The top-right section is for monitor control. By default, the mix of all selected sources is sent to the monitor out and two headphone outs. The latter have both quarter-inch TRS jack and mini-jack connectors, and for headphone channel 1 only that's is a TRRS type, so a headset's mic can appear on channel 1 — potentially useful for gamers wanting to use voice effects. Both headphone channels have a switchable Mix Minus facility to avoid hearing a distracting echo. A stereo six-segment LED meter indicates the level of the Streaming mix, and a Cue button previews this separate mix over the monitors/headphones. Bottom right are six sound pads. These are pre-loaded with radio-show friendly sounds (canned applause etc), but you can record over them using the Shift and Rec buttons, or edit/load samples using the app.

Driving Range

I discovered that you *can* plug the AG08 into a Mac and use it as a multi-channel audio interface straight away, but you really shouldn't: for the AG08 to be properly recognised by the OS and by the AG08 Control app, you must first install the dedicated USB drivers. My M1 Mac's overzealous security system made this trickier than it should have been, but the AG08 manual helpfully explained how to overcome this problem.

When installed, the AG08 presents to the computer as four separate devices. The first, 'Yamaha AG08 DAW', is an 8-in/2-out interface, and provides access to all the hardware input channels, with the stereo DAW output fed back at unity gain to the AG08's monitor, mix, Streaming and aux outputs; you'd choose this one for typical music-production projects. The other three, which can be accessed by other apps at the same time, are stereo. 'Yamaha AG08 Streaming / CH3-4' uses the AG08's Streaming bus as its input signal, so you'd use this to feed a signal to your streaming or podcast recording app. Its output appears on the AG08 as the USB source for channels 3+4. 'Yamaha AG08 Voice / CH5-6' uses only the first two AG08 input channels as input sources, while 'Yamaha AG08 AUX / CH7-8' accesses the main mix. The outputs from these are the AG08's USB source for channels 5+6 and 7+8, respectively. »

On the analogue side, as well as having two mono and three stereo inputs that cater for a range of different sources, the AG08 boasts multiple outputs, on XLRs, quarter-inch jacks and TRRS mini-jack.

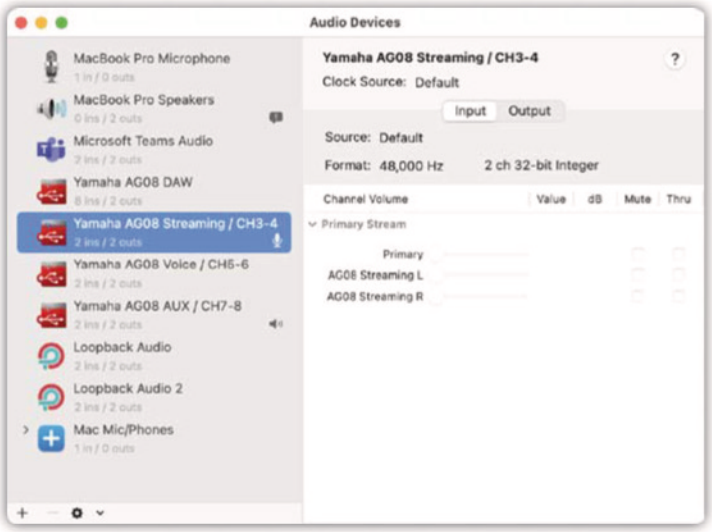
» The reason for all these virtual devices is to allow multiple apps to send different signals to the hardware on different channels. You can, for example, create a mix of your voice coming in through a mic, the sound from a game on your computer, plus music from a browser on the same machine, and route the result to a streaming or recording app, again running on the same machine. Or you could record a live radio-style podcast, with two people speaking into mics, and phone calls coming in on Zoom and Skype, and another app providing music, while the result is recorded to your DAW or streamed by OBS.

Control App

You can do all of that using only the hardware controls but most people will want to use the control app too. This communicates bidirectionally with the AG08 over USB and detects the hardware automatically. (If it doesn't, it'll run in demo mode, implying a need to check the USB connection or perhaps whether your OS's security settings have prevented proper installation). There are two main modes: the beginner-friendly Simple mode is about as far from intimidating as is possible, while Detail mode offers access to all the AG08's digital processing, effects and routing. In both modes, a Sound Check wizard can help you set appropriate levels. This works well, though a small 'gotcha' is that if you've pressed the Cue button in you won't hear the playback test until you set channels 3+4 to Streaming mode or disengage Cue.

In Detail mode, a main window displays key information for all input and output channels. That's already more detail than on the hardware, but buttons then open various windows with more controls. For example, in the main window, you can tweak the compressor threshold and ratio using nodes on a graph-style display, but click the pop-out icon for the compressor and you'll find slider controls for threshold, ratio, attack, release and make-up gain, along with an auto make-up gain toggle, and a choice of soft, medium or hard knee shapes. Similarly, for the EQ the main page has a graph with frequency/gain nodes for each band, but in the dedicated page you can change the filter type for the top (bell, shelf, low-pass) and bottom (bell, high-pass; there's already a dedicated high-pass filter) bands, and each band has sliders for frequency, Q and gain.

Separate windows for channel 1 and 2's effects also provide more options. For channel 1's Voice Changer, you have finer control over pitch and formant and are also treated to a second Voice Effect option: any one of Dual Pitch (great for 'Uptown Funk' style 'duh doooo do' backing vocals!); a band-passed, slightly distorted Radio Voice; a Tremolo; and Pitch Fix, an Auto-Tune style pitch-corrector. Meanwhile, channel 2's amp sim has amp-style gain, EQ and level controls; there may be better plug-in models out there, but it doesn't sound bad. It can kick out some hiss when not in use, though, and I'd quite have liked the option to insert a noise gate or expander; consider this a firmware update suggestion! There are more options for the delay/reverb send effects too: hall, room or stage types, reverb and pre-delay times, high-frequency damping and diffusion, and high/low-pass filtering. At the top of channels 1 and 2 in the main window, a preset menu has useful options for various voices, mics and creative effects but, strangely, you can't save your own presets here. You can save the current channel 1 settings to the hardware's preset pads, of course, but a proper librarian that allowed you to assign software presets to these pads would be welcome.



Once you've installed the drivers, the AG08 is seen by your computer as four separate devices — a multitrack interface for your DAW, and three stereo devices that you can use with multiple apps when streaming or podcasting.

The stereo channels are less tweakable but usefully all have a ducker that can be switched on/off in the main page, or configured more deeply in a dedicated window. They can each be ducked by channel 1, channel 2 or both mics, and you can set the threshold (very useful to avoid breaths close to the mic triggering ducking briefly!), the range, and decay (ie. release time).

For the Streaming output, there's the Maximizer — a multiband compressor that can help to get vocals up to a more broadcast-appropriate level when streaming, though I probably wouldn't record through it. Its 'tweaking' window allows you to set to the crossover frequencies, and the threshold, ratio, gain and attack and release times for each band.

Back on the main Detail page, you can specify which signal plays over the monitor output and another page, accessed via a Settings menu, repeats these options and enables you to choose the source (Streaming or Aux) for channel 7+8's TRRS output, and to link the mix out to the monitor out. This Settings menu accesses various other housekeeping functions too, including firmware updates, footswitch and Mix Minus configuration, a file librarian for the sound pads, and the ability to store and recall snapshots of the mixer's digital settings. Speaking of sound pads, a Pad Settings window, accessible in both modes, allows you to load and trim samples, and to choose the playback mode (one shot, hold, or loop), set the level and the send level to a reverb, and to refine the reverb. You can also choose the pads' backlight colours.

In Use

In use, the AG08 worked very well. I found basic operation super easy, and in terms both of the analogue electronics and digital effects the sound quality is good. Selecting the DAW driver as I usually would for any interface, I was quickly using it for music recording, capturing my vocal through channel 1 and DI'ing a guitar on channel 2, and monitoring the DAW mix on headphones; the headphone amps seemed clean, with plenty of level on tap.

The real *raisons d'être* of the AG08, though, are live streaming and podcast recording. To test this, I recorded several videos in OBS, in which I captured footage from my webcam and various software screens, while using the AG08 to mix the

While the hardware works as a standalone mixer, the AG08 is intended for use with a computer: the AG08 Controller app provides access to more DSP functions.

audio going into to OBS, with some sources being ducked by my mic on channel 1. It was all trivially easy to set up. Another test involved sending the Streaming output to my Android phone, assigning the Streaming signal to channel 7+8's TRRS output. While I'd obviously have preferred proper Android support over USB or perhaps a wireless feed of some sort, this did the job.

When it comes to the processors and effects, there perhaps isn't quite the range found on some competing devices, but the EQ and compressor sound decent and are easy to tweak, the delay/reverb is useful, and the pitch-based effects can be a lot of fun. The 'hidden' effects accessed through the app allow some really fun sounds to be created and assigned to the preset buttons too. What you can't do, though, is have effects inserted on channels 1 and 2 for streaming and simultaneously record a dry signal using the DAW driver — you can do that without reverb or delay, but if the compressor, EQ, Voice Changer or amp sim are used, they'll be printed.

Verdict

What the AG08 does, it does very well. The multiple USB drivers make streaming a mix of external and in-the-box sources a breeze, the hardware sounds good and both the hardware and the control app are super easy to use. Even in the app's Detail mode you're never more than a click away from the main screen. There are some things the AG08 doesn't offer that its main competition (eg. the RodeCaster Pro, Tascam Mixcast 4 and Zoom PodTrak P8) does. There's no built-in recording, for example, and with only two mic inputs you'd need to add external preamps for 'four-around-a-table' podcast recordings. Also, there's no dedicated provision for 'back channel' communication: unless muted, the mics always feed the streaming bus, so you can't have a presenter on one mic and a producer offering prompts on the other. And there's the lack of Bluetooth connectivity I mentioned above.

While that's quite a list, the importance of those features very much depends on what you want a device like this for. I can't think of a podcasting/streaming device that offers quite the same feature set. Its screen-free ease of operation will win many fans, and the app offers more control when needed. It's also worth noting that, as so often with Yamaha, the 'street' price seems to be significantly lower than the official one published on this page! ■■■

£ £738 including VAT.
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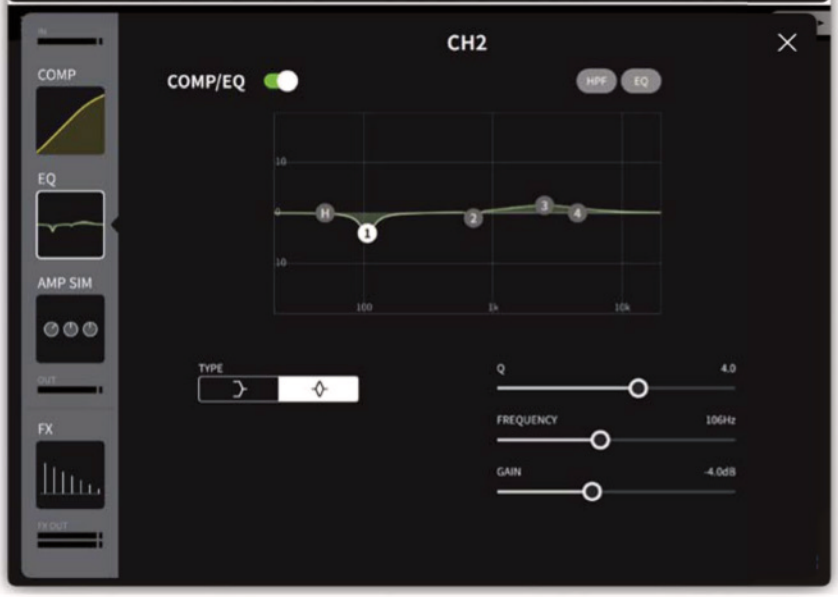


DPA 2012 & 2015

Small-diaphragm Capacitor Microphones

These live-oriented pencil mics offer excellent performance at a very attractive price.

MIKE CROFTS

Talk to almost any sound designer who makes their living in live theatre or large live concerts and shows, and it probably won't be long before DPA mics are mentioned in some context or other. Their product line has become widely known and respected for high quality, reliability and physically robust build, and a new model is invariably a source of professional interest. In this case we're looking at not one but two new microphones, both primarily aimed at the live sound market but equally at home in studio or location recording setups.

Look & Feel

The 2012 and 2015 are pretty much identical in appearance apart from the model number, which can be found on the side of the body down near the XLR end. They are very compact mics, and although light in weight, they do have a kind of substantial feel to them. The supplied clips add a bit more heft.

You can buy both types of mic singly, although I suspect that most purchasers will go for matched pairs, and these come in attractive and strong moulded zipped cases with DPA branding. 'Matched pair', in the case of these and other DPA 2000-series mics, means that their sensitivities are within ±1.5dB. Whether buying one or a pair, each mic comes with a foam windshield and a rather wonderful mic holder. To describe it as a clip isn't really doing it justice, as it's a two-piece compression screw clamp (model UA0961 to be precise) that holds the mic body securely so that it can't possibly come loose no matter how it's angled.

The 2012 and 2015 mics are designed to be tough, both in terms of their physical construction and ability to handle very high sound pressure levels. The wisdom of the past used to regard capacitor mics as more sensitive than dynamics (which they generally are) and therefore more delicate (which these are definitely not). The outer body tube is quite thick and there's a nice little cutaway drawing on the DPA website showing how they are built; in terms of what you can safely put them next to, both models will happily handle SPLs in excess of 150dB (156dB for the 2012, 153dB for the 2015), so pretty much anything on a live stage is on the menu.

Pickup Patterns

Let's get back to the main difference between the 2012 and 2015 mics, namely the pickup pattern. The 2012 is a cardioid, but DPA have focused on achieving a very consistent off-axis response across the frequency range, which translates to less feedback issues on a live stage. A 'normal' cardioid mic would typically be used to capture single sources where there are other instruments fairly close by, and positioning the mic a few inches from the desired source should result in good separation to enable a controlled mix. The smooth, almost flat response of the 2012 gives a great starting point without you having to tame or correct what's going into the desk, and makes it easy to use on just about any instrument. The SPL handling means that it can be right in the bell of a trombone, or as close as comfortable (for the player) to the sweet spot of anything with strings.

The 2015 is a wide-cardioid microphone, and as such has a broader area of capture. It's easy to fall into the

The cardioid DPA 2012 is put to work on acoustic guitar.

trap of assuming that the 2015 should therefore be placed at a greater distance from the source than the cardioid 2012, but that's not necessarily the case: the 2015 works very well as a close mic, but its wider capture angle makes it more forgiving of mic placement, and it has a natural open sound much like an omnidirectional mic but with enough rejection to be useful in a live context. Personally I love using wide cardioid mics as spot mics when recording or just reinforcing larger groups, especially on sections of similar instruments, and they work equally well on a live stage — mind you, they will pick up balance issues within those sections. On single, less directional instruments such as piano, larger strings like cello, bass and acoustic guitar, and almost all woodwind instruments, a wide cardioid is an ideal live mic even if placed up close.

Both the 2012 and 2015 boast a flat frequency response with a gentle lift around 10kHz; the 2012 peak is just north of 10kHz while that of the 2015 is just below, and the 2015 has a slightly more restricted HF response, which you'd expect given its less directional pickup pattern. It goes without saying (but here goes) that both mics have a more than impressive transient response, hence the detailed and faithful sound capture. Despite the differences in polar pattern they can both be used at a range of distances, from traditional ambient stereo pair to indecently close — it all depends on the application, but these are highly flexible tools on a live stage. For live use, self-noise shouldn't ever become an issue especially with a high-quality mic, but the specs below don't give any cause for concern in this department!

Tech Notes

All the relevant specifications are of course on the DPA website, but here are some of the main points. Both mics are based around a 17mm pre-polarised (electret) capsule, and require standard 48V phantom power to operate. They both weigh 83g and measure 94 x 19mm, and unlike some other DPA models they are non-modular designs (you can't remove the capsules from the preamp bodies). Both have the same output impedance of 60Ω.

Where they differ is in their frequency response (40Hz-20kHz for the 2012, 40Hz-18kHz for the 2015, both within ±2dB), their pickup patterns and maximum SPL handling (as mentioned earlier), their sensitivity (8mv/Pa for the 2012, 12mV/Pa for the 2015, ±2dB and measured at 1kHz) and their self-noise (20dBA and 18dBA for the 2012 and 2015, respectively).

Live Action

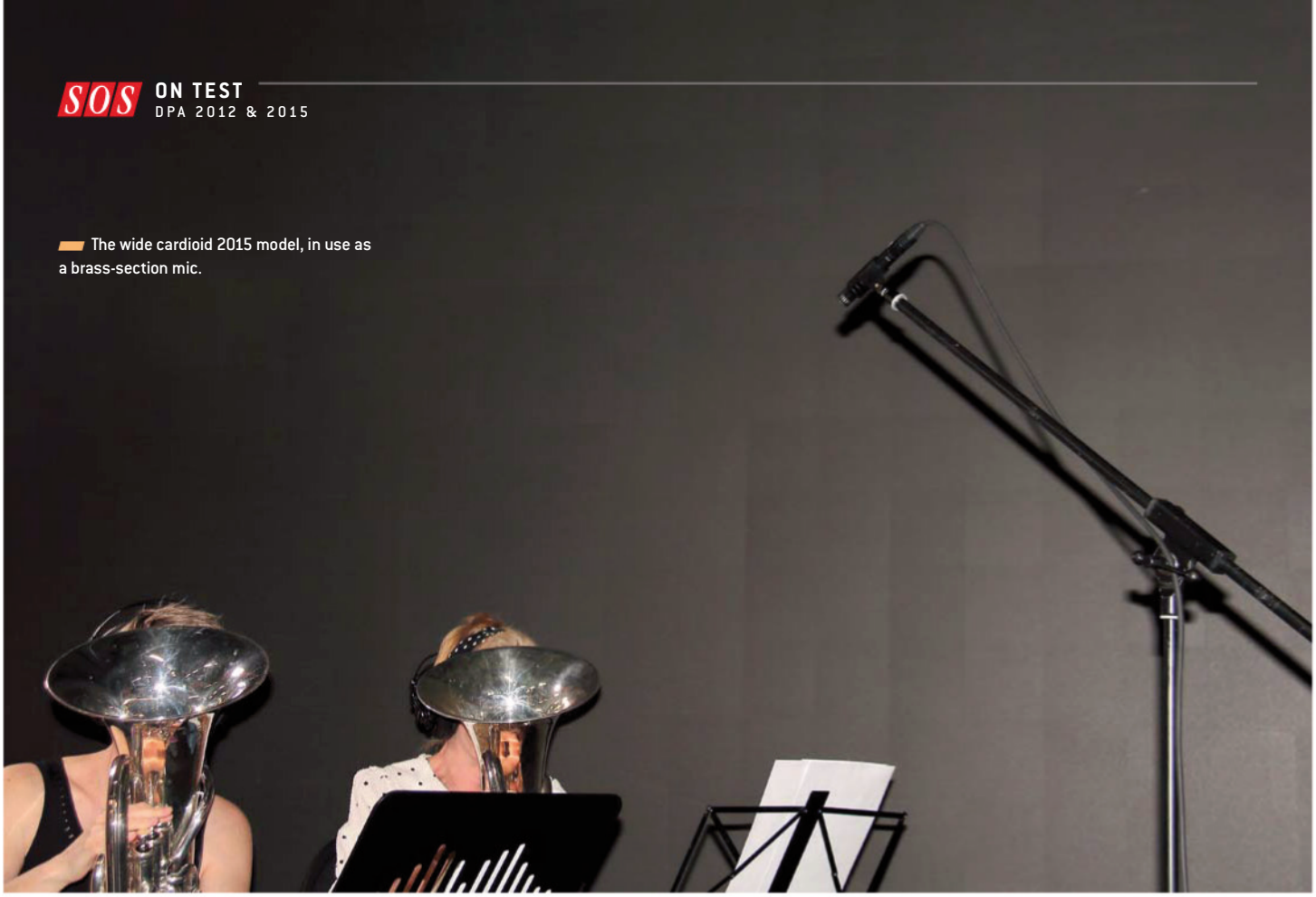
I was sent pairs of both the DPA 2012 and 2015 for review, and they arrived just in time for me to use them at a live club gig featuring acoustic guitars played by John Doyle and Chris Quinn. I decided to use the 2012 pair instead of my usual cardioid capacitor mics and, as we were filming the show, the DPAs scored an immediate advantage with their small size and non-reflective black finish. In soundcheck we experimented with different positions »

DPA 2012 & 2015
£648 Each

PROS
- Sturdy, compact and camera-friendly.
- Reliable, excellent results on a range of sources.
- Natural sound requiring minimal EQ.
- High-quality accessories.

CONS
- None.

SUMMARY
These classy mics sound great, are easy to work with and come in at a much lower price than you'd expect. Highly recommended.



The wide cardioid 2015 model, in use as a brass-section mic.

» including using both mics on one guitar, but in the end a single mic was considered to be less fussy visually and gave exactly the sound we wanted: highly articulate, faithful to the unamplified guitar sound from a few feet away, and reasonably forgiving of the artist moving around a bit during the songs.

I'd describe the overall experience with the 2012s as 'easy', and there was no need for any corrective EQ other than deciding where to set the HPF in terms of the room acoustics, and a slight downward HF slope so as to maintain a nice natural tone. The mics gave a lively and dynamic sound with no impression of compression or constraint of the guitars' dynamics, and this was confirmed when I listened back to my direct-from-desk recording the following day. These are lovely-sounding mics: clear as a bell, great dynamics, articulate and with an open, natural top end.

Top Brass

That same week I was supposed to be making a live recording of a larger band with a 12-piece brass section. For various reasons, this turned into a 'capture-by-sections' session with trumpets, trombones, and upward-pointing weapons including alto horns (tenor horns in the brass band world) and euphoniums. The sections were to play through whole numbers using click tracks and I needed to capture both the whole section with a full, fat sound, and also to emphasise the lead instruments within the section without losing the sense of ensemble — so the 2015s proved to be ideal, with one positioned centrally to the section a couple of feet back and the second one closer to the lead player.

Again, the word that comes to mind is 'easy', as they just worked exactly as intended, delivering a lovely broad, natural 'in the room' sound, and the closer mic was so easy to blend into the mix. The following week I used them in a live rehearsal, again recording different sections but also listening live in the hall. The 2015 is an ideal mic for bringing out a section of a larger ensemble, whether it be a trio of horns or a whole drum kit. The only thing I didn't get the opportunity to try the mics on was vocals, but I suspect that the 2015 in particular would be just as impressive on, say, a group of two or three BVs.

And speaking of drums, both these mics are now available within a drum set called the DPA DDK4000, which comprises a single 2012, a pair of 2015s, a 4055 kick mic and three 4099s.

Practically Speaking

Setting up for a live show isn't always done under ideal conditions, and one thing I've learnt over the years is that, apart from a few specialist mics, my preferred models are always those that are more forgiving. Sometimes you just don't have the opportunity to spend time fine-tuning and ironing out frequency foibles or proximity problems. For me, this is precisely where the DPA 2012 and 2015 come into their own — having used them a few times, I'd feel confident deploying them virtually anywhere and would pretty much know what to expect.

The 2012 and 2015 are more affordable than some of DPA's high-end mics, but they offer all the trademark DPA qualities and bear the brand name with complete justification. These are serious, high-quality microphones but at an achievable price point for anyone looking to own something from the DPA stable. Much more than that, though, they simply do a more than excellent job wherever you'd use a cardioid or wide cardioid capacitor mic on a live stage. Both mics exhibit a wonderfully clear, natural sound with great articulation and presence, and although they are designed principally for live sound, they would of course do the same job in a recording environment.

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ANN MINCIELI

Producer, Engineer & Studio Owner

Photo: Cheryl Fleming Photography

Jungle City Studios occupies both penthouses in a split Manhattan tower block. This is the Penthouse West live room.

Photos: Ramon Rivas @ramonrecords

After rising through the vibrant New York studio scene of the ’90s, Ann Mincieli developed a 25-year partnership with the biggest R&B star of the 21st Century, and now has her own state-of-the-art Manhattan studio.

SAM INGLIS

“She’s the female version of Prince!” says Ann Mincieli of Alicia Keys. “If you listen to the production on her records, it’s cross-genre, You might hear a distorted Wurli. You might hear a distorted Moog, and John Mayer comes up to her and goes, ‘Hey, how did you make that guitar tone? It sounds just like Prince on “Purple Rain”!’ and she goes: ‘It’s not a guitar. It’s my Moog going through a distortion pedal and an amp.’

“I realised that the type of engineer I wanted to be, where we’re painting a sonic picture on each record, was the type of producer Alicia was. So that’s why we started to work together, and grow together. I wanted to be constantly evolving, getting into the depths of engineering.”

No Formula

Keys and Mincieli have been growing together across more than two decades and an incredible amount of recorded music — far more than has ever seen the light of day, in fact. “We start with a blank piece of paper for each album, and for some albums we do 240 songs, 90 songs. Each album is a journey, and sometimes it evolves so that by the end it’s completely different than what her vision was, but that’s the beauty of her.

“You sell millions and millions of records and you’re the biggest-selling R&B act of the millennium, but also you want to continue to grow musically. I feel like years ago, artists did that more, like your David Bowies and your Led Zeppelins. Now it’s just like this formula where you’re told: this won’t work on streaming, you got to do a radio edit, you can’t have a guitar solo any more, it won’t get played on radio. It takes away from what we really subconsciously love about music, which is musicians getting in a room and just playing and making magic.

“We had a good guitar player in recently and he’s like, ‘But I played the first hook. Can’t you just copy and paste it to the other hooks?’ and she’s like, ‘No, we want you to play it down, that way we get the human element and feel of how you play. It is not the same each time.’ Your eight-bar loop might be an eight-bar loop, but it drifts within the eight bars. Every bar might be a little bit of a different tempo and that’s fine. That’s how she works.

“She has no rules, which is the beauty of it. Ninety-nine percent of the time, she’ll never take a track that’s pulled up on a hard drive that someone has brought in. it’s about making these songs from scratch and collaborating, and you got to be able to play with her and rock with her. So, a lot of times you do need musical chops. You can’t just be copying and pasting and fake your way through it because she loves to arrange, if you listen to her songs. On two songs we worked on with Emeli Sandé, called ‘Not Even The King’ and ‘101’, those are one take. What we captured — even the delay throws — is the take that’s going on the record. So I have to make sure I have these magnet mics by DPA in the piano and I can close the lid. And the bleed is controlled, I can have some control »

She Is The Music

Like many successful engineers and producers, Ann Mincieli takes her responsibility to help and mentor other talented people very seriously. As well as recruiting young engineers for Jungle City and other studios around the world, she’s also set up a non-profit dedicated to helping women succeed in the business. “Myself, Alicia Keys, Jody Gerson and Sam Kirby decided to make an organisation about helping women in all facets of the music industry. So check that out: sheisthemusic.org. We wanted to flip the numbers and every week and every day we’re pushing and creating opportunities for women in our space at a global level.”

Ann Mincieli has worked closely with Alicia Keys for more than two decades.

» over my piano mics. My EQ on them is that the voice ain't bleeding into them!"

Freedom

A part of Mincieli's role with Keys is as an enabler, creating an environment that allows the artist to go wherever inspiration takes her. "She has 15 keyboards set up, no matter where we work around the world. We go to Peter Gabriel's studio [*Real World*] in Bath and spend three weeks there. We ship all our gear, take a day to set up, we have two assistants that travel with us. She has her own Pro Tools rig that loads up all her sounds and all of her virtual synths, and I have my rig; this way, she can load all her sounds and I could be loading a song up. We don't have to clash, we can be working independently. Then all her libraries aren't on my rig, taking up all the space — because the sound library is huge. And that's how we roll.

"And she has old keyboards from Mellotrons to Jupiter-8s... We might not travel with those, but we definitely will travel with the newer version of a Mellotron. We might take our Juno-106s, our Juno-60s, all of our Dave Smith Prophet Six, Prophet 10, Prophet X, all of that stuff. We load everything up on a console. We route it into my Pro Tools. And then she sits and programs by herself. She uses the [*Akai*] MPC, she uses [*Native Instruments*] Battery, and she sits and she programs and sometimes she's in a room by herself. We leave and she has the freedom to write in there, so she walks around and is playing and is experimenting. And then when she needs us, she'll say 'Hey I want to record this!'

"She used to sequence a lot of her keyboards in her Pro Tools rig. But now she's like, 'I just play this one time and it's recorded. Why do I even need a sequencer?' There's something about her swing even that these sequencers can't mimic, even if she wants a quantise. So she just plays her stuff down.

"And then there are times where she starts off on a piano, and she plays to a click. And her tempo drifts even though you're playing to a click, right? She's a human. So I have to then go in and mark every bar and make the click track rise and fall. And her piano is really dictating the tempo. Then the drummer will come in and he'll play to the same swing because the click will rise and fall.

The control room at Jungle City Penthouse East. Visible lower right is the studio's vintage EMI TG-series console.

Every bar. When it becomes a problem is if I don't make a tempo map. Then the Pro Tools click is just staying on one tempo, and the feel of the music is slightly off. So that's why it's important for me to make a tempo map."

Life School

Keeping up with such a freewheeling, varied approach to production requires some pretty serious engineering chops, which Ann Mincieli gained the old-fashioned way. "I started at a studio named Marathon, and then a studio named Skyline in New York, which was really a popular space in place in the early '90s. I started as an intern, and I worked there for nine months, and I would work there on weekends because I was so young — I wasn't even graduated at high school. When I did graduate I was what we call the general assistant at the time. I worked my way up from being an intern in 1993; I started there and then at Right Track Recording, which was an amazing facility. And I got my whole foundation at those studios in '93 and '94.

"I went on to go work with François Kevorkian at Axis Studios, which was a big turning point for me, because he was very on to what was coming next. I was learning Pro Tools, which was Sound Tools back then, and Logic Audio, which was a MIDI sequencer back then when no-one knew what it was. I tried to pick all these stepping stones for myself.

»

The Key To Perfect Vocal Takes

No matter what genre Alicia Keys is pursuing, one thing that's always front and centre is her voice. Asked if she has a set way of tracking this unique instrument, Ann Mincieli's answer is thoughtful. "She uses several mics depending on the vibe and the sonic of the song. For starters we use a [*Shure*] SM7; no matter where we are around the world, you can put that mic up at any keyboard setup and she can get ideas out, and obviously there's no bleed. If you solo an SM7 vocal track, you won't hear much bleed, but you do have to raise your mic pre up high. It's a dynamic mic and I think the magnetic field is very small on it, but it's amazing. And you saw everyone, from Stevie Wonder to Michael Jackson, on those mics back in the day. I use the foam filter and keep everything flat.

"I also use a vintage [*Telefunken ELA M*] 251, and then I have an M16 that was designed by Telefunken USA. It's a $1200 mic and I know there's many variations of it right now, but we have a prototype that they made. She uses that as well. And there's similar microphones out there that are 1200 bucks till this day, and it's an amazing mic. And we use a [*Sony*] C800G.

"She loves textures, so she might use a different mic on backgrounds for textures, and a different mic on her lead. I think it depends on the key of her song; if she's in a key and it's pushing and it's squeezing her tone, then I might not be on her M16. I just have to figure out the range and the key of the song. I've had her on a [*Shure*] Beta 58 in the control room, like she'll be singing, and vibing because she's writing, and then I wound up keeping some of that as a take. I see a lot of artists doing that since the pandemic, too. Sometimes you have to roll with it. It's less about like, 'Hey, this might not be proper right now!' but you know what? It's working."

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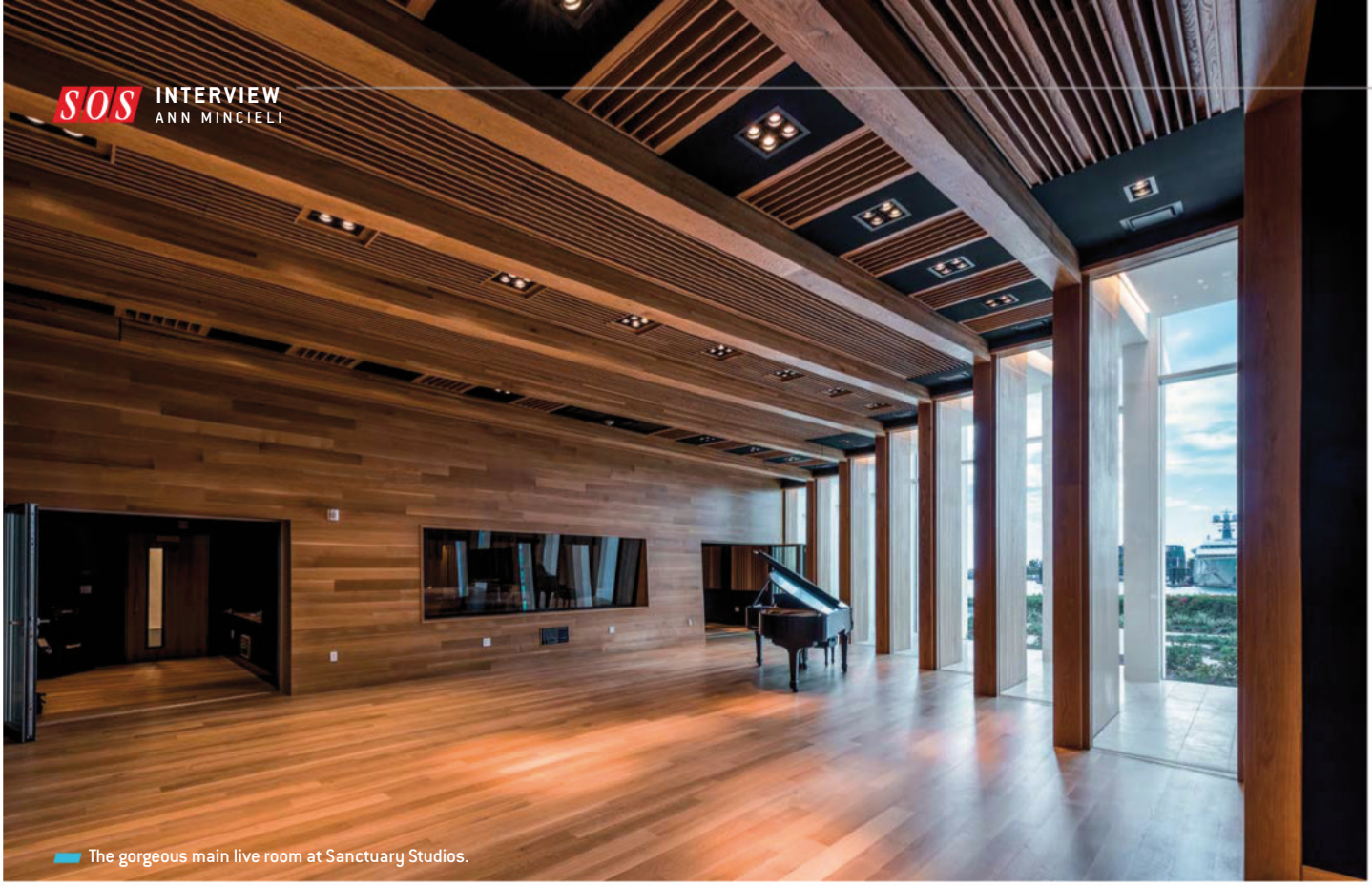


The gorgeous main live room at Sanctuary Studios.

Photo: Cheryl Fleming Photography

» "I went to Quad Studios in '95, became a staff engineer, and I was there for seven years, working with some of the biggest artists there from Metallica to Bon Jovi to Blue Man Group to Mariah Carey. I learned a lot from her engineer, Dana Jon Chappelle, and Tony Maserati, and I really figured out what I wanted to do in the industry. And I learned about the art of engineering. Quad was an important place; they were the first studio to have the 9000 J-Series console from SSL. So, I feel like I grew up on Solid State Logic. We went to England for four weeks and learned this new SSL because it was such a learning curve from the G and the G+. So yeah, that's kind of my history — and Quad was where I met a young writer named Alicia Keys, in 1998.

"We had a producer named David Collins at Quad who was using Cubase and Logic and Pro Tools way before anyone else. And then I worked a lot with Luther Vandross and Courtney Love. I was with Michael Beinhorn on this nine-month session where we were transferring vocals in and out of Pro Tools way before anyone knew what Pro Tools was. And I really learned about bit depth and bits and sample rates and clocking."

Be Good

The young Ann Mincieli's drive to succeed didn't just push her to learn new tools and techniques. She also understood the value of going the extra mile for clients. "I really worked hard at just trying to be better than the rest. I played sports my whole life and I was very competitive and I always felt like if you can outwork and out-compete and outperform, you're not gonna have much competition. At Quad we had seven rooms. We had 13 assistants. But I always did the little things, and clients would request me. And every time a big artist would come in, everyone would fight for the gigs, but I would get the gig — because you have to be good at the little things.

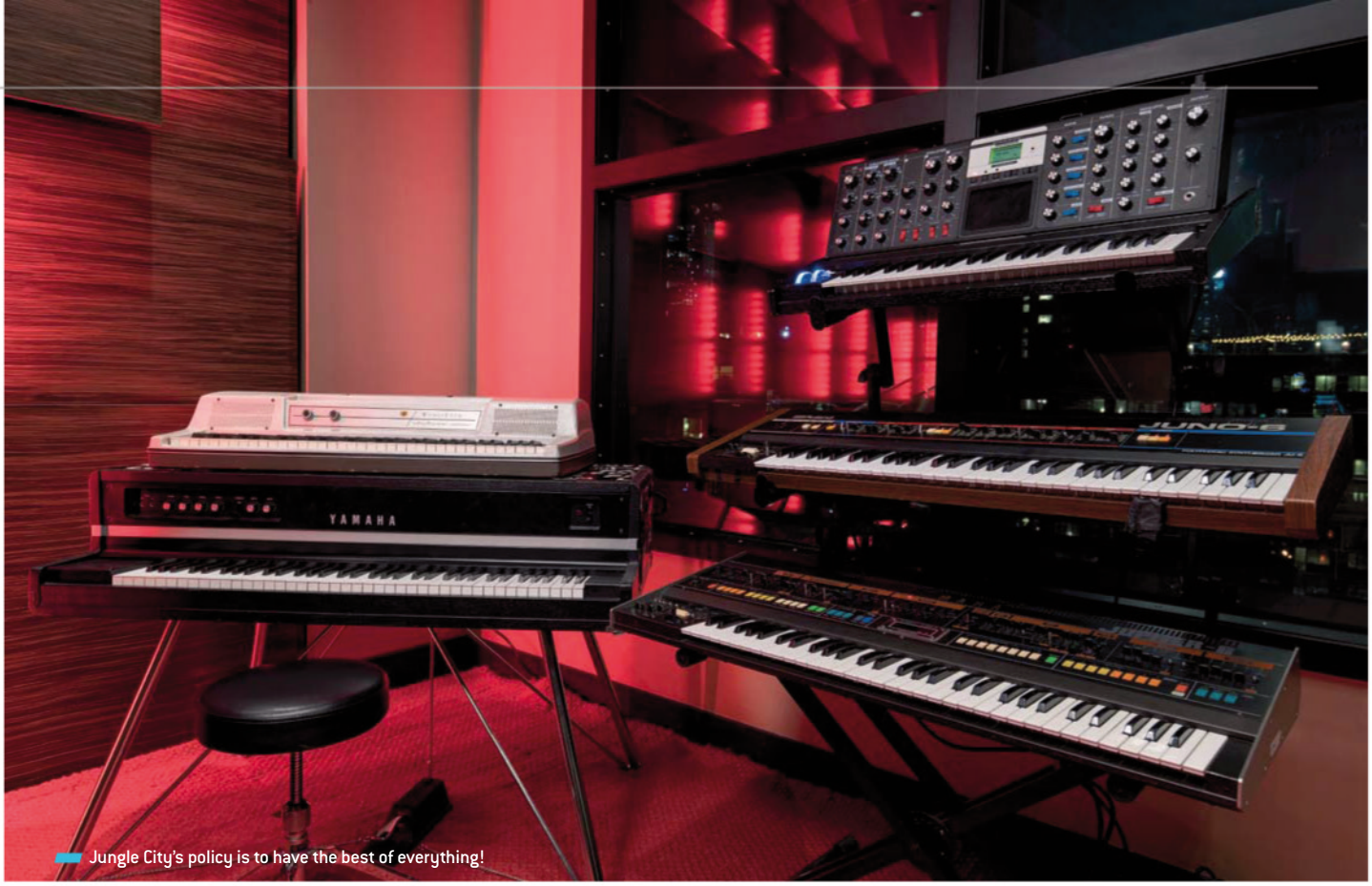
"I would be able to pull up a whole recall from Tony Maserati, use my ears, and recall every channel. Sometimes there's 40 or 50 pieces of gear, and there's no plug-ins, right? There's a hundred-input SSL you are recalling. So there's lots of notes. I learned a lot from François Kevorkian and his studio, because a lot of mixing went on there, and he had really great assistants who documented that stuff — and documented, back in the day, was writing it down. Insert points? What was on the busses? What was on the sends? The SSL took your recall, but then it was up to you to write all the information. What are your calibrations on your DATs? What are your calibrations on your half-inches? What are your calibrations on your 3348, right? And I would be able to pull those recalls up even from other studios when they rented gear. An LA-2A at Quad and an LA-2A from Right Track might be two different-sounding units. Even if you have all the notes, you got to be able to use your ears, listen to the effects, the returns: is it loud enough? So I learned all that, and was able to use that to my advantage.

"I always remembered a studio manager telling me: be so good that they're gonna request you when they come back. And Quad had a lot of repeated customers, and I would get requested, sometimes by two sessions at once. So, that was my mentality: just be good and it won't matter if you're male, if you're female, if you're young, if you're old, whatever it is won't matter."

Welcome To The Jungle

Many of the great New York studios, including some of those in which Ann Mincieli learned her trade, are now closed, victims of a real-estate boom and the downturn that hit the recording business in the 2000s. So it raised eyebrows when, in 2010, Mincieli decided to build a high-end studio in Manhattan. "It was at a time where the industry hit a low. All the studios were closing, streaming was taking over, everyone didn't know where the future of the industry was going to go. And here I was in full-blown construction mode!"

It was a brave call, but against some expectations, Jungle City has flourished. Indeed, the whole studio industry is now much healthier than appeared possible at that time, and Mincieli is often called



Jungle City's policy is to have the best of everything!

upon as a consultant for other high-end projects. Amongst other things, she's Studio Director of the stunning Sanctuary Studios in the Bahamas, and has worked with Sony to develop in-house studios in key cities such as New York, LA and Nashville.

Mincieli realised that when New York studios were going under in the early 2000s, it wasn't simply a matter of rising rents and falling income. Often, their business model had left them tied into long-term financial arrangements that left them vulnerable. "The business plans behind a lot of the old studios was where I didn't want to make the mistake, which was leasing a space for 30 or 40 years and not owning it, which was leasing consoles and gear and not owning it. And then it takes you 30 years to pay off a lease, and then you don't even own it. I didn't want to do that. I bought my gear with cash.

"My family invested with me, which was my sister and her partner. They both worked at Met Life at the time, and my sister found the location, which 13 years later is in the heart of Hudson Yards and the High Line. It's not in the middle of Times Square. It's not in the middle of tourist land. There's a lot of art galleries around.

"I wanted everything about my studio to throw you back to what studios really were. I wanted the staff, the seven-star hotel type mentality, that I learned from Rose from Record Plant and Troy Germano and Hit Factory. I really wanted to create the modern-day Motown. I didn't want to take any shortcuts. It started with the location and it being a place for inspiration. I could have leased a cheap space and put 10 studios in it, but I wanted quality over quantity. I wanted to be a destination."

The Learning Curve

Mincieli sees the current renaissance of the studio business as an opportunity for some young engineers to receive the kind of comprehensive, on-the-job training that she got back in the early '90s. She recognises the value in colleges and universities that teach recording and production, but says that education alone can't equip someone to be a great engineer. "You need to learn protocols, you need to learn technically what it takes to run a session and a facility, You're not getting all of that. The schools do a great job of changing the curriculums, because kids coming out now even at some of these high-end schools, they don't want to learn an instrument, they want to learn Ableton and they want to learn how to be a better DJ. It's changing before our eyes, and a lot of them come out now as producers.

"And at some of the studios now, it's very easy to set up, right? You can just set a microphone up, route the music to the headphones, and that's the most you're gonna learn. I go to schools and they want to learn Auto-Tune. They want to learn Melodyne. And it's like: you need to learn the whole foundation of how to build a house before you put the shingles on! There's a whole foundation that you're not learning, and I believe in the depth of engineering. And when you work at Jungle, there's many different types of sessions that are coming in. You're gonna have many different types of encounters. Whether it's Depeche Mode, setting up, three days of modular synths, to another client bringing in two-inch tape and locking up tape machines. It's a depth to the engineering and assisting roles.

"I work on an initiative, too, with Sony Music. Rob Stringer put a committee together to have studios for each label and in each flagship location. I put those studios together for Sony, and it's a way to keep the community going. How are kids coming out of school gonna be able to learn if they're just in their bedrooms? They need to connect to be part of our industry. So I'm glad that more facilities are opening. I'm part of the hiring process in some of the locations, and I'm really proud to hire people and keep the community going. So that's my goal: to keep the music studio community going.

"Being great means a lot of discipline. I took my career really seriously when I was young. When some of those guys were out partying, I was at home going to bed because I had a session next day. I was so passionate about this industry, and I still am to this day. I'm married to this industry. It's all I know and love.

IN THE ROUND

Spatial Audio For Live Performance

Immersive audio is a powerful way of presenting live music — and it's becoming more and more accessible to independent artists.

HALINA RICE

When I delivered my first immersive set in 2019, the addition of spatial audio was not something that had even crossed my mind. At the time I was gripped only by the desire to create a show uniting electronic music with visuals — creating something to take an audience out of their day-to-day experience — and was working with designer Jan Petyrek to achieve this.

Jan and I were searching for venues to test out our nascent AV set when we were offered a last-minute slot at a Hackathon event at Abbey Road Studio 2, the aim of which was to encourage participants and performers to innovate. The company providing the sound system, L-Acoustics, explained they were bringing a 12.1 system into the venue, and suggested I might like to program the set for this setup. This seemed intriguing, and with encouragement and guidance from their engineers I was able to reformulate my set using their L-ISA software in just a couple of days.

Live For Live

Like many electronic music producers, I use Ableton Live for both production and performance. For my live sets I was already used to bouncing audio of each track into component stems such as kick, percussion, bass, effects and synths, and placing these into a number of scenes in Ableton, with each scene triggering a whole track. I would then add a variety of instruments and samplers into the Ableton set to play live over the audio playback.

The first time working in L-ISA introduced me to the now-familiar process of stemming my tracks in a much more granular way to create audio objects for spatial mixes. In brief, spatial audio gives you the opportunity to place and animate all of these audio objects, aka sound sources, independently within the auditorium. An L-ISA Processor — either physical or virtual — maps the objects to the physical speaker array to convey the spatial mix you have created. The programming to achieve a spatial mix involves adding metadata to each created audio object. (For more about the principles of object-based mixing, see our introduction to immersive audio in the January 2022 issue: www.soundonsound.com/techniques/introduction-immersive-audio.)

Metadata can include information about panning, elevation, and even effects such as distance created through the addition of dynamic reverbs. These can all be changed over time, either using snapshots within the software engine or — as felt more comfortable in my workflow — using L-ISA plug-ins in Ableton Live and drawing in automation. These then communicate with the L-ISA Controller, which integrates information for all your audio objects, speaker setups and monitoring options and also includes a spatial effects engine.

These days, I might create between 10 and 20 stems depending on the project. The idea is to achieve a good balance between the number of outputs and programming required, and the flexibility to treat sounds differently. As an example, if you bounced a synth and a vocal sample together in the same stem, they would receive the same metadata and would be placed in the same way around the room, which may not be the desired effect. However, you can group multiple granular stems in the L-ISA software if you actively want them to receive the same metadata; for example, you could group a number of synths together and place them around the room in one go.

Advance Planning

For that first project at Abbey Road, the main challenge was monitoring, and this

Photo: Joe Baker

is still a key issue for the development of immersive sound today. At that time, the only way to audition my programming was within a multi-speaker monitoring setup, and I completed much of the object-based mix at the L-Acoustics Creations showroom studio in Highgate with an 18.1.5 studio speaker array. In the words of Ferris Bueller: "It is so choice. If you have the means, I highly recommend picking one up." But that luxury isn't often available, and many people need a more affordable and portable way to test out their panning and effects choices.

In 2021 the launch of the L-ISA Studio suite brought the addition of binaural headphone monitoring, a feature also available in other popular spatial audio software such as the Dolby Atmos Renderer. Binaural monitoring uses psychoacoustic techniques to emulate how we perceive sounds coming from behind or above us. There is also the option to use a head tracker, which allows a more fully realised rendition of a 360-degree space, constantly reinterpreting the sound as you turn your head. In my experience, binaural monitoring has never sounded exactly like the eventual room setup, but it gives a clear enough idea of the movement of sound objects to enable me to program a set using headphones. Even for reproduction in stereo, a venue can sound radically different from a studio mix and require some adjustment, so I don't think spatial audio brings any new challenges here.

That initial, experimental Hackathon opened a door for me to consider using spatial audio in future shows, adding to the palette of options with which to create immersive experiences. It had particular relevance to my practice, as the compositions ranged from pure sound design — delivered all around the audience — to beat-driven IDM tracks, which benefit from a static frontal image (kick, bass and so on) enhanced with effects and synths placed around the auditorium. I was excited to push forward — and then Covid happened, and these ambitions were put on hold.

Multi-format

By the end of 2021, the world was opening up and live performance became a possibility again. In the intervening time, I had been building up the visual aspect of my set, incorporating generative art and elements of mixed reality. In addition, I had developed my amateurish skills in programming lights, and was keen to put all these elements together with spatial audio to create a coherent immersive show. But where and how could such a show be presented?

The author's setup uses Ableton Live and L-ISA from L-Acoustics.

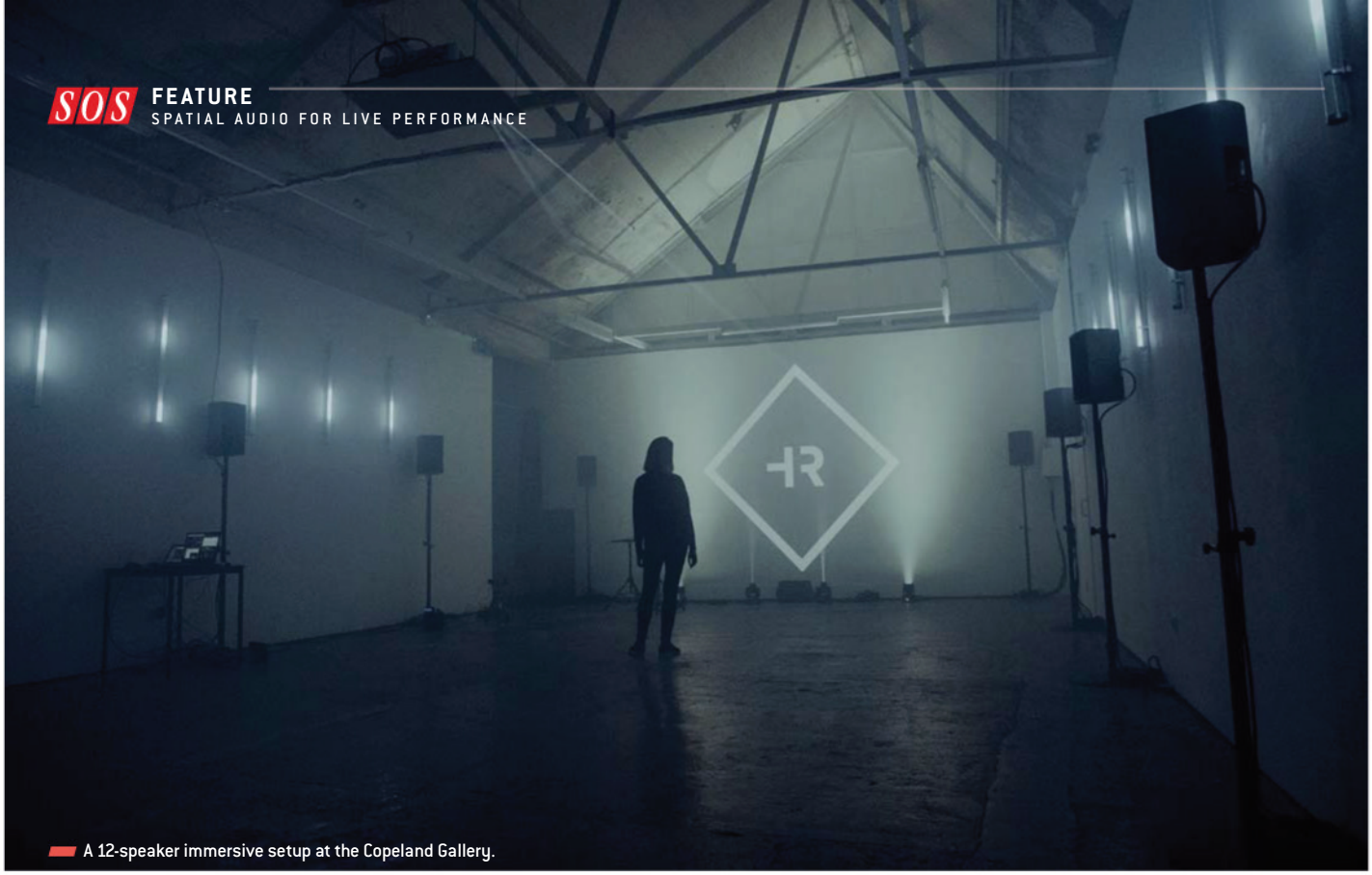
Photo: Chloe Hashemi

Although there are some entry-level venues that have spatial sound systems, and more are slowly emerging, the number is still quite small. There are good reasons for this: it's more expensive than stereo to fit and maintain, and as immersive audio is still a niche area for most artists and genres, there is rarely a requirement on the venues to have such systems, nor is there necessarily the content available. Audiences still have minimal familiarity with immersive sound and are content to receive audio in stereo or even mono, even though their experience may be enhanced when presented in spatial audio. With these factors in mind, venue operators are naturally hesitant as to whether a spatial system would pay back the investment in time and money.

A good example of a forward-thinking, emerging space is Dareshack in Bristol, where co-founder and creative director Adda Cohn has pioneered the implementation of a d&b Soundscape system into a 200-capacity venue so that grassroots AV groups can trial immersive content. Unfortunately, this is still a rarity, so if you want to put on an immersive live show, you may well find yourself needing to hire in a system.

For my first post-pandemic show, I was fortunate enough to have some support from Arts Council England and so took the route of hiring both venue and sound system. A 12-speaker L-Acoustics system was installed for three days at the Copeland Gallery to implement both a daytime installation piece and two

The L-ISA plug-ins allow spatial audio metadata to be written directly into a Live session.



A 12-speaker immersive setup at the Copeland Gallery.

Photo: Michael Augustini

» evening performances. The job of fitting the audio, projections and lighting into the venue seemed for a while to be overwhelming, but the audience reaction was so intensely enthusiastic that the general indication was that there was something in this formula to pursue.

Down To EartH

Following this experiment, I was keen to find other venues where I could present the immersive show. The natural progression in London was to take this to EartH Hackney: a 670-capacity venue built in a reclaimed theatre, where a 17.1.8 in-house L-Acoustics L-ISA configuration made it easy to integrate my existing set, and the L-ISA-trained in-house sound engineers ensured the sound had great delivery on the day. The main consideration in preparing my set for this space was to do with the layout of the venue and the system. At the Copeland Gallery, we had had a true 360-degree setup with identical speakers placed evenly around the space. By contrast, the setup at EartH is more akin to the arrangement you might find in an arena, with the majority of speaker power along the front of the stage and supplementary speakers to the sides, back and ceiling of the auditorium. This still allowed enough scope for some playful positioning and movement of sounds across the audience, and again the reception was very enthusiastic.

The EartH show highlighted another benefit of an immersive system: the ability to achieve coverage in a larger space. This was also evident at a performance I did at a church in Berlin: an unusual though beautifully designed round space where half the audience were on the ground floor and the other half on a mezzanine level. For this show, a 16-speaker system was brought in with speakers encircling both levels. In this instance, spatial audio was an effective tool for a Heineken style effect of reaching the parts other setups could not reach.

Lost In Translation

Just because a venue is equipped for spatial audio doesn't meant their system will necessarily integrate so easily with the software you've used to program a show. As an example, I performed in 2022 at arts venue Iklectik, also in London, which has a bespoke multi-speaker setup created by the organisation Amoenus using Max for Live. Working with their in-house engineer, we decided that rather than reprogram my entire set for the venue, we would use Dante Virtual Soundcard to send speaker signals coming from my virtual L-ISA Processor in the L-ISA Studio software out to the front of house for dissemination into their speaker array. This method was used again to drive a 7.1 speaker system in Milan at the Triennale Theatre.

This was a relatively simple solution, but it raises a wider point about future interoperability of systems. Will there ever be a common language or standard for translation between spatial audio software and processors? Many artists release their recorded music immersively using the Dolby Atmos format, but may then have to alter or reprogram for live spatial audio where systems such as L-ISA Studio by L-Acoustics or Soundscape by d&b are more often found. In theory, the beauty of object-based mixing is that it can elegantly translate to any speaker setup, whether that be in a cinema, at home, in a venue or on headphones. As far as I know, however, »

Did They Enjoy It?

One of the hardest things to do in evaluating audience feedback is to isolate the influence of spatial audio on their response. Most of the time the impact of the sound, or the way in which sound is delivered, is harder for attendees to articulate than, say, the size of the projector screen, or how the lights are mounted around the room. Sometimes I receive very specific descriptive feedback — I have heard people enthuse about the quality of the sound separation they experienced — but these are quite professionalised terms and often come from those who have worked previously in music production. More usually the feedback is just that "The sound was great!", or there's a vague sense of enhancement compared to conventional shows. I also have the impression that the audience feels anticipation and excitement just from the physical presence of speakers all around the auditorium. In the case of the Copeland Gallery, it even seemed to enhance the general aesthetics of the room. It also contributes to the audience's sense that the show is designed for and around them, and is therefore a different experience from other music events.

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EarthH in London is a slightly more conventional performance space with a stage and a well-defined 'front'.

Photo: Michael Augustini

» there is currently no simple way to take a mix from one object-based immersive format to another.

Outside The Box

Whilst traditional music venues have been relatively cautious about installing immersive systems, other spaces have been a bit more proactive. In London, it seems as though a new immersive space or exhibition appears every week. Some theatres and other cultural venues have been quick to adopt this as a future trend and are responding to changing audience demands. Earlier this year I worked with The Old Market theatre in Brighton, to look at a phased evolution of their layout to present immersive events. Using a modular system of screens plus a number of high-end projectors they can change their main audience area into a 'white cube' space ready for 360-degree projection and audio. I created an 'in the round' version of my show using projection-mapping to show graphics on all four of the white screen walls. For sound we brought in a 12-speaker L-Acoustics system evenly spread around the space, with three speakers mounted at the top of each wall. Beanbags were placed in the space so the audience could be even more involved in the performance.

The programming for this show was interesting and a little challenging, as there was not even a nominal 'front' to refer to. I found myself replicating core elements such as kicks and bass on speakers on opposite walls to give a sense of evenness for the fundamentals of beat-driven tracks. However, it was also possible to be totally experimental with other elements.

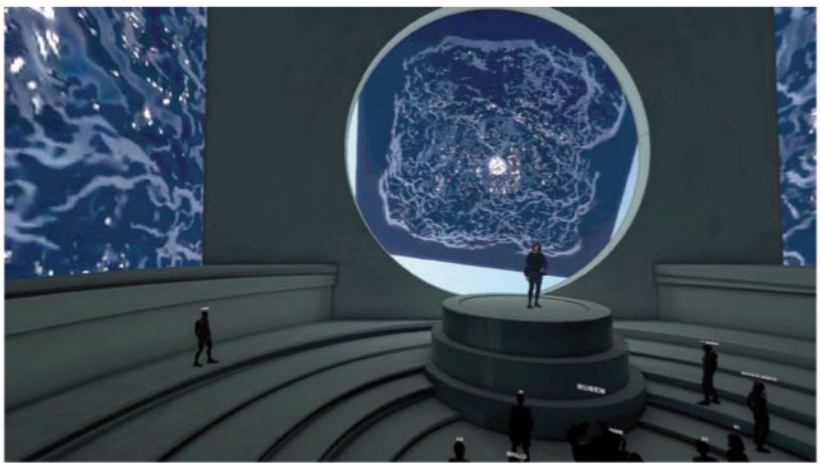
Visual designer Freny Antony and I worked together to create bespoke graphics that moved around the four walls of the room, and the sound programming worked in tandem with these to create the illusion of audiovisual objects moving in space. The possibilities of this seem very interesting for future AV shows, and I can imagine that there will be companies working on products that can integrate programming of the two elements together in the future.

The venue took detailed audience feedback for this event to understand the

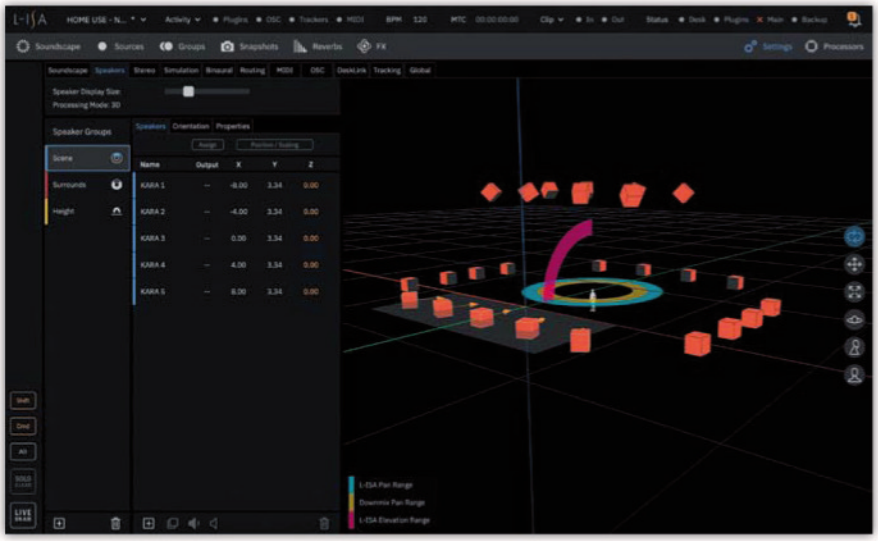
Free Space

Physical spatial audio setups can be costly, but there are also opportunities for immersive audio where the speakers are not physical at all. Metaverse events have started to become more accessible for independent artists. In February this year I worked with Bristol-based Condense Reality, which enables content creators to stream live 3D video directly into online platforms to create a metaverse event. This included a volumetric recording of me as the performer with the audience joining as avatars. It was presented as a live binaural mix to the audience. This was an interesting first step for me of uniting a virtual environment with spatial audio. The complete freedom to programme sound here is again quite disorientating as there is no physical constraint on how and where sound should be placed. In this instance, as it would be received as a binaural mix mainly in headphones, there was a practical consideration to mono and centre bass and kick, allowing other elements to exist in a wider field.

Working with doctoral candidate and researcher Victoria Fucci the resulting assets have been recreated in a VR application for headset use and will be used as a basis to research the effect of audio and visual mixing conditions on emotional experiences for music events in VR. An interesting field of development and one where spatial audio need not present as much of a barrier in terms of cost and setup time.



An immersive metaverse performance with Condense Reality.



The EartH speaker setup modelled in L-ISA.

A 360-degree immersive show at The Old Market theatre in Brighton, March 2022, with 12 speakers mounted above screens that enclose the entire space.

Photo: Chloe Hashemi

A 'crowd-reactive' installation with 70+ loudspeakers at The Outernet, London.

Photo: The Outernet

impact, and again, this was overwhelmingly positive, indicating a strong desire for more similar events. Looking forward, as venues such as The Old Market evolve their sound systems to become immersive, there could be a vacuum of content and potentially a requirement for more producers to create in immersive sound for these spaces.

Along the same lines, there are also opportunities emerging in retail environments as brands capitalise on the rise of interest in immersive experiences and look to create 'activations' with an element of spatial audio. An interesting space utilising this is The Outernet in London, which is billed as a concept in retail, entertainment and tech and incorporates enormous LED screens and a 70+ multi-speaker spatial audio system. Passers-by can stop in and engage with their programme of cultural and brand-related content.

I was invited to create the sound design for Monolith, a crowd-reactive piece created by visual designer Jack Dartford and production company Chaos Inc. An immersive L-Acoustics L-ISA system was installed in the space, but had not previously been used for its spatial capabilities. It was a very enjoyable experience to create reactive spatial sound on this huge canvas, with the installation reaching over 100,000 passers-by across the course of a week. The likelihood is that more similar spaces, both fixed and 'pop-up', will emerge as brands enable investment in spatial audio.

Space Exploration

As an independent artist, I am still encountering practical challenges when it comes to presenting shows in spatial sound. The limited number of spaces with spatial audio setups means I have to retain the ability to perform in stereo for now, but adoption by venues and practitioners is increasing, and the tools for programming and monitoring spatial audio sets are becoming more accessible. Study programmes on spatial audio are appearing on music production curricula everywhere, and I think this will be a completely familiar format for the next generation of producers.

Overall, I'm excited to see the development of immersive sound, and to enjoy the relatively rule-free environment where one can pioneer and trial new creative ideas. Whatever the future holds for this tech, it's something we can all be involved with to shape and create new audience experiences.

Boss SDE-3000D

Dual Delay Pedal

This pedal brings Roland's classic '80s rackmount delay into the 21st Century, with dual delay engines and modern control options.

PAUL WHITE

Whereas many of the early digital delays sounded somewhat sterile, Roland's rackmount SDE-3000, released in 1983, soon gained a reputation for sounding warm and musical and became a staple in the armoury of many pro musicians. In part, that was down to the analogue circuitry that sat around the digital delay section, and in part to technical imperfections, including a slightly imprecise master clock that added a subtle modulation to the sound. A unique tonal character was imparted by the converters available at the time, which I believe were just 12-bit, and the SDE-3000 also included an LFO-controlled modulation section, had the ability to store presets, and boasted switchable delay polarity and (for then) a long maximum delay time. Even now, used models are highly revered. But what once took up a lot of space and cost a lot of money has now been packed into the relatively compact confines of a new Boss pedal, called the SDE-3000D. In fact, it offers rather more than the original mono rack unit...

Boss SDE-3000D
£449

PROS
- Authentic recreation of the vintage SDE-3000 sound.
- Dual engines can be deployed in a number of mono and stereo configurations.
- Comprehensive MIDI control.

CONS
- Only the amount of pedalboard real-estate needed.

SUMMARY
The SDE-3000D brings all the practical and sonic benefits of the original in a dual delay pedal with additional features to meet modern needs.

Thanks For The Memory

The SDE-3000D houses two delays, each with independent parameters and separate panning options, and has a mono/stereo input and output. The modulation and tone-shaping aspects of the original have been extended too, and there are 100 preset memories. The front panel gives direct access to two banks of four presets, though the footswitches can be reconfigured to access all 100 presets.

The stereo inputs and outputs offer all the signal routing permutations of mono and stereo, and comprehensive MIDI support makes the pedal compatible with modern needs. The two delays themselves can be configured either serially or in parallel by selecting the appropriate Delay Structure, which includes the option to set up as two mono delays or as separate delay and dry outputs. For those wishing to emulate Eddie Van Halen's three-amp 'wet/dry/wet' setup, there's an alternative version of the pedal called the SDE-3000EVH, and in this one each delay and the dry signal are treated to separate output jacks — the idea is that the dry sound feeds its own amp in the centre, with the two delays being sent to their own amps on either side, to create a wide stereo delay with no intermodulation distortion of the dry sound. The EVH also has an effects loop, with separate send and return jacks.

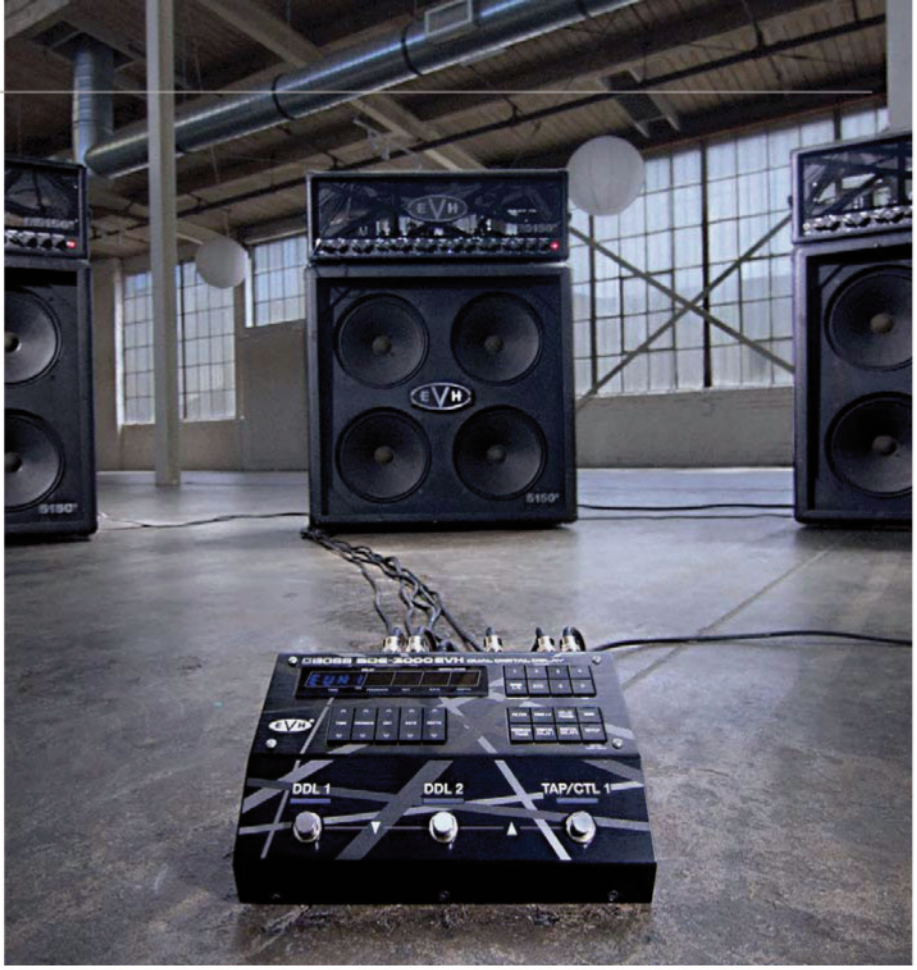
The delay time can be switched in two ranges, maxing out at either 1.5 or 3.0 seconds, and both the imperfections in the clock of the original and the sonic signature of the original analogue circuitry have been carefully replicated using DSP, to recreate the familiar warm, organic sound. Phase (polarity) switches are provided for both the primary delay and the delay feedback loop, along with variable low-cut and high-cut filters for the delay feedback, and each delay has a separate modulation section.

Boss have given the pedal a retro-style look with a control panel reminiscent of the original SDE-3000, complete with seven-segment LED displays. Buttons are used to access the parameters — there are no rotary encoders in sight — with the main delay and mod values shown in the display as with the original so there's no menu diving necessary for general operation. In Play mode, the display shows the delay time, the feedback value, the delay level, mod rate and mod depth; additional information is displayed when

in Setup mode. Each of the delay and mod parameter sections has its own up/down button for setting the value. A further 16 buttons look after the remaining functions including Setup mode, polarity switching, filtering, 2x delay time and preset storage.

On entering Setup mode, the Time button is used to move through the various parameters, with the Depth button used to change values. Once in Setup mode there's lot you can access, such as the functions of external switches or expression pedals, MIDI setup and so on, so having the manual to hand is pretty much essential, though in most cases once you've set things up the way you want them, you shouldn't need to venture there very often — all the important delay parameters are right there on the front panel.

The right-most of the three footswitches (Tap/Ctrl 1) is assignable and can be set to operate tap tempo, hold, and other functions, though using the expansion control facility makes operating additional functions during performance more practical. TRS jacks accommodate (up to four) footswitches, two expression pedals, or a Boss GA-FC or GA-FC EX foot controller. The left-most pair of foot switches access Delay 1 and Delay 2, though these can be reconfigured for preset access. To save space, MIDI is on 3.5mm TRS jacks located on the side panel — these are commonly used for MIDI in modular synth setups but an adaptor cable is needed to interface with devices that use standard 5-pin MIDI connectors. A USB-C socket, also on the side, allows for direct connection to a computer (macOS, Windows or iOS) for MIDI — a Boss driver must be downloaded and installed on your computer before this can be used. Power comes from an included in-line 9V adaptor. MIDI can be used to recall up to 100 presets, to adjust parameters using MIDI CCs, and for sync.



The SDE-3000EVH version of the pedal features separate outputs for the dry signal and each delay line, to allow you to recreate Eddie Van Halen's wet/dry/wet three-amp setup.

Impressions

It has to be said that the SDE-3000D is rather special. But whether or not you need the functionality that the SDE-3000D offers depends very much on how you use delay live and how much pedalboard space you're prepared to give over to it, especially if you'd want to use extra switches or expression pedals to get the most out of it. You might prefer something more straightforward or favour the sound of emulated tape or BBD delays, and in that case you might be better off checking out Boss's other delays (from their compact DD-series pedals upwards, they have almost as many delays as the UK rail network!). But if delay is a key part of your sound, and especially if you work with stereo delays, the SDE-3000D won't fail to impress: you might just have found your new best friend.

It works perfectly well in mono, but in stereo it really comes alive, and there's that EVH version for anyone wanting to use it with a three-amplifier rig. As with the original hardware, it has enough of an inherent character to avoid sounding clinical, with the modulation section adding a lovely sense of movement. It also offers a huge range of delay times, from less than one millisecond up to three seconds, making it possible also to recreate phasing and flanging effects, as well as the more usual delay/echo effects — and having two delay engines means you could, for example, set up chorus and delay at the same time. In other words, you might find that this pedal can cover the functions of two or three of your existing pedals.

The bottom line is that if you coveted one of the original SDE-3000 machines, you can now have two of them in a much more compact format and with enhanced functionality, and all for rather less than you might pay for a beaten-up, 40-year-old original!

On the rear panel, alongside the mono/stereo inputs and outputs, there are jacks for external control options.

£ SDE-3000D £449. SDE-3000EVH £539. Prices include VAT.
W www.boss.info/uk

Neumann MT 48

USB & RAVENNA Audio Interface

SAM INGLIS

Neumann have joined forces with Merging to create an interface that's both excellent and accessible.

Neumann are deservedly famous for their microphones, but over the years, they've manufactured other types of audio gear too. In the '70s, their range encompassed mastering lathes, mixing consoles and outboard equipment. More recently, parent company Sennheiser bought Klein+Hummel, leading to the development of Neumann-branded monitor speakers. And the subject of this review is another collaborative product, which has brought about another corporate acquisition. Neumann began talking to Merging Technology about developing a Neumann-branded audio interface more than two years ago, and the relationship blossomed to such an extent that Merging are now also part of the Sennheiser group.

Origin Story

The MT 48 takes as its starting point the Merging Anubis, a product that has, thanks to its unusual flexibility, enjoyed the distinction of being reviewed twice in *SOS*. The Anubis can be transformed to suit different roles by loading Missions. The Monitor Mission configures it as a hugely powerful stereo and surround monitor controller, while an Anubis running the Music Mission becomes an equally versatile audio interface for tracking and mixing.

The Anubis itself is a compact, desktop interface, but unlike most such devices, it connects to the host machine over Ethernet, using the RAVENNA protocol. This enables one of its key features: 'peering' with other Merging products such as the Horus and Hapi. The additional I/O is integrated in a completely seamless way, such that every aspect of the peered device can be controlled as if it was built in to the Anubis. This elegant approach to I/O expansion is beautifully handled by Anubis's other stand-out feature, the high-quality touchscreen that offers complete control over its internal mixer and other settings. For the full low-down on the Anubis Music Mission and peering, take a look at our review from June 2021: www.soundonsound.com/reviews/merging-anubis-music-mission.

A New Anubis

The Merging Anubis remains a current product, and as far as I'm aware there are no plans to cease development or roll it into the Neumann line. The MT 48 occupies the same metal shell, albeit in a different colour scheme and with a Neumann logo in place of the Merging branding. It also has the same quotient of analogue I/O as the Anubis: there are two quarter-inch instrument inputs and two headphone

Neumann MT 48
£1750

PROS

- Inherits almost all of the Merging Anubis' features, including its excellent touchscreen implementation and ability to operate and peer as a RAVENNA interface.
- Adds easy-to-use, plug-and-play USB connectivity and optical I/O.
- Sounds superb.

CONS

- Only offers a single optical in and out.
- No equivalent of the Anubis Monitor Mission, and hence no surround support at present.

SUMMARY

Neumann and Merging have successfully tweaked the feature set of the existing Anubis to nudge it further towards the music production market. They haven't fixed anything that wasn't broken, but the addition of USB connectivity is a big plus.

outputs on the front, while the rear panel features a pair of mic/line inputs on combi jack/XLR sockets, a pair of main outputs on XLRs, and an additional pair of balanced line outputs. Another pair of 'general purpose' quarter-inch sockets can be used to connect footswitches, or as MIDI I/O with optional adaptors.

Look to the left of the back panel, though, and you'll spot some crucial differences compared with the Anubis. The MT 48 can operate as a network interface, but has only a single RJ45 Ethernet port. All current Anubis variants have two, as a redundant connection is required in markets such as broadcast. In place of the second RJ45, the MT 48 features two USB Type C sockets. One of these is labelled Power, but as the MT 48 draws up to 5A, bus powering won't usually be an option; it comes with its own USB supply, or can be powered from a suitably hefty power bank. Alternatively, it can be powered over Ethernet.

The other USB socket is labelled Data, and signals that this is not only a RAVENNA interface. The MT 48 is a class-compliant USB device, and as such, works with Apple computers and iPads without the need to install a driver. Windows users will of course need to install an ASIO driver for low-latency operation.

Peering over RAVENNA is supported just as it is on the Anubis, whichever protocol is used to connect the MT 48 to the host computer, but unlike the Anubis, the MT 48 also sports a pair of Toslink optical digital ports. These can carry stereo S/PDIF digital audio or up to eight channels of ADAT digital audio in either direction. High sample rates are supported, with the ADAT channel count falling to four at 88.1 or 96 kHz, and two at higher rates. The highest sample rate available on the MT 48 is 192kHz; the Premium version of the Anubis, by contrast, offers 384kHz PCM and DSD recording too.

Hands On

One of the drawbacks of networked audio is complexity, and setting up an Anubis or MT 48 system over RAVENNA using the full-fat ANEMAN utility is not for the faint-hearted. Merging have simplified this process significantly with their UNITE configuration wizard, but it's probably still fair to say that audio over IP is both overkill for and intimidating to many people. The provision of a USB alternative thus makes perfect sense and helps to differentiate the MT 48 from its progenitor. Since networked operation was described in our reviews of the Anubis, I'm going to concentrate here on the MT 48 as a USB device.

On macOS, the MT 48 uses the built-in Core Audio driver, but you can and probably should also install the MT 48 Toolkit app. This runs in the background and places an icon in the toolbar. Clicking on it brings up a short list of options, of which the most important are Open Web Control and Launch Remote Control App. These actually do pretty much the same thing; like the Anubis and some other interfaces like MOTU's AVB range, the MT 48 is controlled using the hypertext protocol, meaning that its control panel is essentially a web page. The Remote Control option simply opens this in a separate, dedicated browser app rather than in your default web browser.

The browser-based control panel is no more elegant or graphically sophisticated than it was with the Anubis, but it's clear and functional. More to the point, it's largely superfluous, since the MT 48 has the same superb touchscreen implementation as the Anubis. I've come across several interfaces with touchscreens now, and this is the only one that offers comprehensive, intuitive control over absolutely every parameter. I'm sure most users will rarely open the control panel software and will work entirely from the interface itself.

As on the Anubis, both the front panel and the browser-based control panel give full control over the internal mixer and also over the MT 48's mic preamps. These are exactly the same as those of the Anubis, but the gain parameters are presented slightly differently. Whereas the Anubis offered -12dB pad and +12dB boost options alongside the fully variable, digitally controlled 0-66 dB gain range, the MT 48 is configured so that the default gain range is 12-78 dB, with the option to introduce either -12 or -24 dB pads. There are a handful of other minor changes, such as new graphics for the gain controls and a wider choice of cutoff frequencies for the high-pass filter, but in essence, the two units are sonically identical. That's a good thing, because it means the MT 48 inherits not only the Anubis's excellent preamps but also its superb audio specs, the highlight being its 136dB output dynamic range.

The Anubis's internal effects and signal conditioning are available here, too, meaning that input channels have comprehensive EQ and dynamics plus a send to a global reverb. What's new and exclusive to the MT 48 is that both the raw and the conditioned signals are available separately as inputs to your DAW. In other words, you can give the artist a compressed, equalised version of their sound in the cue mix, and record either or both that and the dry signal to separate tracks. In fact, since there are four onboard mixes available, you could fade up the processed signals in the artist's cue mix and the dry ones in the engineer's mix. The EQ and dynamics are »

At the back we find a pair of USB-C ports, an RJ45 Ethernet connector, optical I/O ports, 'general purpose' quarter-inch I/O for footswitches or MIDI, a pair of quarter-inch line outputs, a pair of XLR main outputs, and pair of XLR/jack inputs.

If you're reading this in the print edition then this photo is only a little under life size. Cute, isn't it?

» fairly utilitarian digital processes, so I don't think you'd use them to add character in the way you might analogue outboard, but there are certainly circumstances where this facility could be useful.

Performance

What hasn't changed compared with the Anubis is the general sense of no-expense-spared quality and attention to detail. The shell seems to have been milled from a single piece of metal, making it robust, heavy and a very effective heat sink for the internal circuitry. Small holes at either end reveal the presence of internal fans, but these never came on in my testing; and, like virtually every other aspect of the MT 48's design, from line level calibration to headphone crossfeed to reverb decay time, their operation is adjustable from the touchscreen. The option of powering the MT 48 over USB or Ethernet creates redundancy, so it doesn't bat an eyelid if you plug one connector in and then unplug the other.

The Anubis sticks in my mind as being perhaps the best-sounding interface I've reviewed, and the MT 48 is no different. From its super-clean yet musical preamps to its high-class headphone amps, it's pretty close to state of the art, and I would happily use it even in the most sonically demanding roles such as mastering or classical recording. The addition of USB operation makes it truly plug and play, at least on macOS and iOS, and the optical I/O makes it trivially easy to add a few extra inputs when needed. Having said that, this is a fairly minimal implementation of ADAT expansion: you'd perhaps expect an interface in this price bracket to offer a second pair of optical connectors to maintain the channel count at high sample rates, and a word clock output.

I tested the MT 48 on the same computer I used for my Anubis Music Mission review, and the comparison in terms of latency performance between USB and networked operation is interesting. Signal processing that's implemented on the interface itself inevitably adds to the round-trip latency, and on some interfaces I've tested, this additional delay can amount to several milliseconds. Low latency has been a priority in the design of the MT 48's internal mixer and processing: the exact figure depends on the choice of filter you select for the D-A converters, but can be as low as nine samples. This brings the minimum round-trip latency at 44.1kHz to just under 5ms at the smallest 32-sample buffer size available over USB. That's lower than the 7.2ms you get with the same buffer size under RAVENNA, but can't quite match the 3.6ms available with the RAVENNA-only 16-sample buffer size. Either way, it doesn't quite match the performance of typical Thunderbolt interfaces, but is easily good enough for most purposes.

Conclusion

So, should you buy the MT 48, or the Anubis? There are a few Anubis features missing from the Neumann product; as already noted, these include DSD operation and a second RJ45 port for redundancy. There's also no current equivalent of the Monitor Mission, and the MT 48 lacks the Anubis' ability to import speaker EQ settings from Sonarworks SoundID. But unless you're working in broadcast, doing high-end classical recording or running an Atmos studio, I don't think these features will be missed (and Neumann have promised surround support in the future). By contrast, the addition of plug-and-play USB connection and ADAT expansion definitely make the MT 48 more appealing to the music producer on the go. And, in my humble opinion, the Anubis was already pretty tempting to start with.

£ £1750 including VAT.
W www.neumann.com

www.heritageaudio.com

The sound of yesterday.....
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PSP Stereo Controller 2

Stereo Mix Correctional Plug-in

PSP are well known for their range of effect and processor plug-ins, but their PSP Stereo Controller 2 revisits one of their earliest products. It's a plug-in that's designed specifically to help identify and correct problems with stereo tracks or mixes, whether in L-R or M-S format. Sample rates of up to 384kHz are supported, as are all the common Mac/Windows plug-in platforms, including AAX, and there's Apple Silicon compatibility too. Authorisation is via the iLok License Manager (no physical iLok dongle required) and up to three licences can be activated to the same iLok account.

With the ability to split the audio into separate bands with user-adjustable crossover points, Stereo Controller 2 allows the user to adjust the stereo width separately in the high and low bands. It also offers the ability to flip the polarity of either channel, and to swap the left and right channels. A goniometer scope display at the top of the centre section shows the stereo width and positional balance of the processed signal: if you just see a vertical line you know that your signal is, essentially, mono. Stereo width can be narrowed to mono or widened up to 400 percent — higher settings should obviously be avoided unless the original signal contains very little stereo information!

Two additional bar-style displays show Correlation and Balance. If the correlation meter strays too far to the left, it could indicate that some parts of the mix have opposite left/right polarity, suggesting possible cancellation problems. If over fully to the right, the mix is in mono. The Balance display should also hover around the centre if the left/right balance is nominally even.

Up to two milliseconds of inter-channel delay can be dialled in, to compensate for mic distance errors of up to two feet (68 cm) on a stereo track, while the Stability control allows the left/right balance to be adjusted without affecting the location of centre-panned signals. The Low, High and Main sections have individual Bypass buttons and there's also independent gain adjustment for the frequency bands. The Centre signal has its own pan control, while Balance works as an overall left/right balance control. There's also a global Width control in the middle section which moves the L and R markers in the Scope display when adjusted. It is also possible to drag the L and R markers and the Width control will follow.

There's nothing complicated about Stereo Controller 2, and there's little that you can do with it that couldn't be achieved using other plug-ins in combination. But having all the tools and metering in one place like this does make the job so much easier, and a utility plug-in such as this has many applications, for example narrowing or even making mono the very low frequencies within a mix to aid vinyl cutting or to ensure equal distribution of low frequencies across both speakers. The ability to correct independently any side and centre positional imbalances is extremely useful too, as is having separate control over the stereo width in different parts of the spectrum and the ability to solo the Mid and Sides signals separately. It comes with a useful selection of presets, and while it's unlikely to be your first call when it comes to creating fun sounds, it can be a powerful ally when fine-tuning your final mixes or when processing stereo tracks. *Paul White*

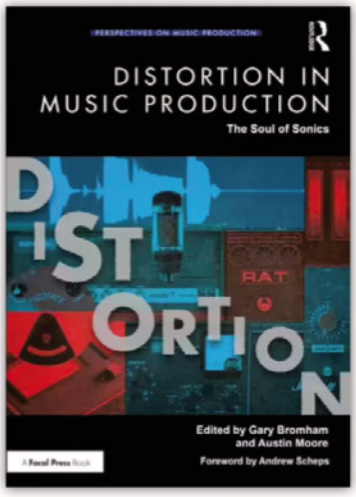
£ $99
W www.pspaudioware.com

Distortion In Music Production – The Soul Of Sonics

Book Review

This 290-page book, published a few months ago, is the latest addition to the Focal Press *Perspectives On Music Production* series, and it focuses on the creation and application of distortion in all its forms as a musical effect. In the interests of transparency, I should mention that yours truly is quoted briefly a couple of times, as are several other *SOS* colleagues and contributors, but that's probably not surprising given the significance of *Sound On Sound* magazine as a source of reliable information in the audio industry!

Edited by Gary Bromham and Austin Moorem and featuring 17 chapters, *Distortion In Music — The Soul Of Sonics* (ISBN 978-0-367-40585-4) is written by 19 different authors and is divided into four sections. These cover the technology, the perception and semantics, retrospective perspectives, and musicology of distortion. So, in simple terms, it's the how, what, when and why of creating and using musically pleasing distortion effects. The book appears to draw heavily on academic research papers and presentations, and although

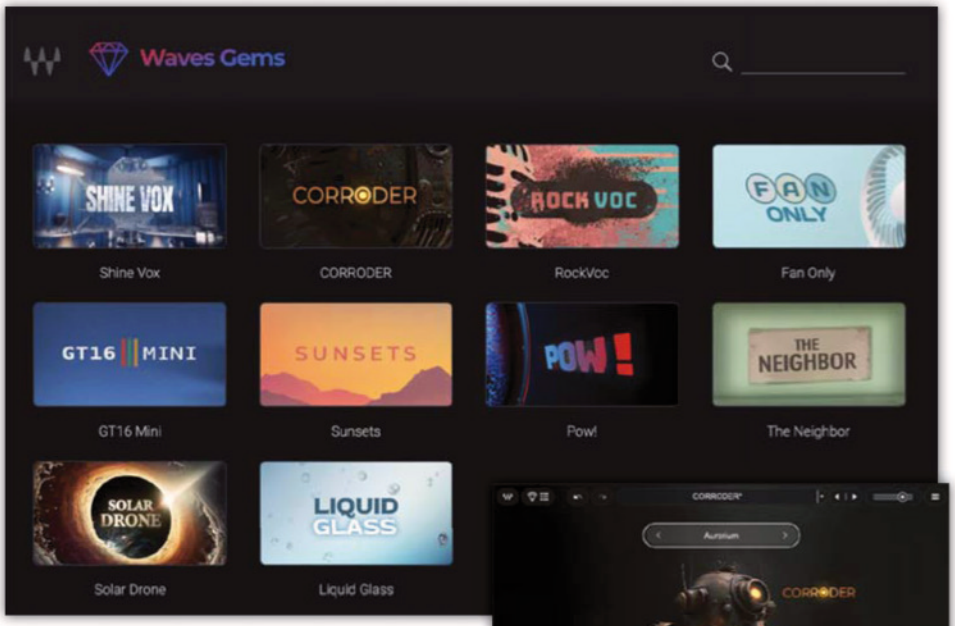


it couldn't be described as a light read it is very readable, and has logically separated chapter topics. I found much of interest throughout, with some fascinating insights, as well as a lot of historical information on the development and use of distortion effects and techniques that were previously unknown to me.

Section 1 covers the technological aspects of distortion in five chapters, including its historical development in analogue circuitry, modern software implementations, and its use as a side-effect of dynamic range processing and in different mic preamp topologies. Section 2 covers the terminology employed to describe different distortion effects and spans three chapters, while section 3 uses four chapters to examine how distortion effects have been used in the past. Finally, the last section's five chapters examine relatively contemporary applications of distortion and the associated aesthetic aspects of its use. Comprehensive references are included at the end of each chapter for further reading opportunities, as might be expected in a book of this type, as well as a full index.

To nitpick slightly, there are a few diagrams that, at least in the hard copy version, were rendered less useful than they should be due to their having been created in colour but rendered in black and white. A few typos here and there made it past the proofing stage too, and there were occasional claims and assertions which raised a sceptical eyebrow in this reader. Nevertheless, taken as a whole this book is informative and interesting, and it presents some enlightening viewpoints on the creation and application of distortion as a musical effect, both historically and as a current production tool.
Hugh Robjohns

£ Paperback or eBook £34.39. Hardback £96.
W www.routledge.com/9780367405854



Waves Gems

Effects Plug-ins

Waves Gems is a collection of creative effects that offer stompbox-like simplicity when it comes to the controls, but each one is nonetheless capable of creating a variety of useful treatments that could take a long time to setup from scratch using other plug-ins. The individual Gems cover a range of use cases, including vocal processing, motion-inducing modulation for instruments, drum treatments and drone processors. Available only as part of the Waves Creative Access subscriptions, five of the Gems come with Waves Essential, and you get all 10 of them if you sign up to Waves Ultimate.

All are accessed through a single plug-in window, which opens with a browser-type screen that shows all of the available Gems. Click on one to launch it (only one Gem can run at a time in each instance of the plug-in) and you are greeted by a relatively small number of presets and just a handful of knobs for parameter adjustment. To get back to the browser view, you just have to click on the Gem icon to the right of the Waves logo. There's an output level control for each Gem, along with the option to resize the window. Vocal treatments include Shine, to add sheen and doubling, or Rock Vox for beefing up and adding colour to rock vocals, this with a delay effect built in.

For lo-fi aficionados, there's Neighbour, which creates the sound of music coming from a neighbouring apartment, and has variable distance, door, hallway and floor contributions, while the delightful Corroder adds some very usable lo-fi grit and dirt.

Meanwhile, G16 Mini serves up bit-reduction effects reminiscent of video games. Pow adds weight and dirt to drum sounds with variable sub-octave, while Fan Only adds a modulation effect that sounds as though your audio is being played through a rotating fan — complete with the wind noise! For the more genteel side of music, Solar Drone uses shimmer, filter, distortion and modulation to create long, lush drones, with the option of pulsating rhythms, while Sunsets adds a languorous texture that's based on delays and modulation plus some filtering. Liquid Glass also uses filtering and modulation, in this case to add the character of a glass full of liquid being sloshed around, which is a rather more musically pleasing effect than it might sound from that description!

Despite their simplicity, these effects can be used to very quickly come up with effective treatments that would fit well into EDM, chillwave and ambient music. You cannot buy Gems separately. Rather, it's being offered as an incentive to buy into one of the Waves subscription packages. Or, of course, you could view it as bonus content for existing subscribers, who've just had a bundle of creative fun added to their plug-in armoury.
Paul White

£ Only available in Waves subscriptions, which start at $12.50 per month.
W www.waves.com

Spotlight

Active Ribbon Microphones

Looking for that smooth ribbon tone with the convenience of a modern mic? Look no further...

LUKE WOOD

Ribbon mics can be a useful addition to any collection, with their characteristically smooth high-frequency response and low-mid emphasis often providing a more flattering sound on sources that can come across as a little harsh when captured with a capacitor mic. Recognising that the low output of traditional passive designs can be quite demanding, often requiring a high-quality preamp capable of providing a significant amount of gain — or the use of in-line boosters — many manufacturers now offer versions fitted with phantom-powered active preamps or buffers to deliver a much hotter output. In this month's Spotlight, we take a look at some of the obvious brands to check out if you're on the lookout for an active ribbon mic.

Sontronics Delta 2

Sontronics

Sontronics currently make three active ribbon microphones, all of which are designed, developed and hand-built by the company in the UK. High-quality components are used throughout, with each model employing a custom-wound transformer manufactured to the company's own design. They also utilise RFI filtering in their circuitry to provide maximum resistance to interference. The Sigma 2 (www.soundonsound.com/reviews/sontronics-sigma-2-delta-2) is an updated version of the company's first ever ribbon design, and is somewhat of an 'all-rounder' that is recommended for use on a wide range of sources from vocals and spoken word to string, woodwind and brass instruments. The Delta 2 is of a similar design, but features a lower sensitivity, a more robust ribbon and more internal protection, allowing it to withstand higher SPLs. The Apollo 2 uses a pair of ribbon motors in a Blumlein configuration, making it possible to capture stereo recordings with a single microphone.

£ Sigma 2 £829, Delta 2 £829, Apollo 2 £1999

W www.sontronics.com

sE Electronics RNR1

sE Electronics

There are two active ribbon mics in sE's range. The RNR1 (www.soundonsound.com/reviews/se-electronics-rnr1) was the first of three microphones developed in a collaboration with legendary audio equipment designer Rupert Neve. The design employs separate transformers for the capsule and output stages, connected via a discrete Class-A circuit that compensates for the ribbon's high-frequency roll-off. This allows the microphone to offer the 'warm' low- and mid-range character associated with ribbon designs, but with an extended high-frequency response. A switchable 100Hz high-pass filter is present, and the RNR1 is capable of withstanding a maximum SPL of 137dB. It also comes supplied with a custom shockmount. The VR2 (www.soundonsound.com/reviews/se-electronics-voodoo-vr1-vr2) also offers a wider frequency response than many ribbon designs, this time using a mechanical diffusor designed by sE's CEO Siwei Zou to extend the response up to 18kHz. It comes supplied with its own custom shockmount, and can withstand SPLs of up to 135dB.

£ RNR1 £1199, VR2 £382

W https://seelectronics.com

Golden Age

Golden Age Project's R1 range offers a selection of mics inspired by RCA's iconic 44, promising to deliver high-end performance and vintage character at an affordable price point. The R1 Active MkIII and the R1 Tube Active are of a similar design,

Golden Age Premier R1A

but the latter uses a 12AX7 valve-based buffer amplifier in place of the former's FET circuitry. The premium Golden Age Premier range includes the R1A (www.soundonsound.com/reviews/golden-age-premier-r1a), which uses high-quality components throughout, with a discrete amplifier joined by a Lundahl 1927A transformer. Output levels across the range are similar to those of many capacitor microphones, meaning that quiet sources can be recorded without high-gain preamps — and with a maximum SPL handling of up to 160dB (165dB for the Tube version), they can capture even the loudest of sources. The R1A and R1 Active MkIII also feature a switchable 160Hz high-pass filter and -10dB pad.

£ R1 Active MkIII £219, R1 Tube Active £339, Premier R1A £519

W www.goldenageproject.com

AEA

AEA A440

AEA's rich heritage dates back to 1976, when they began servicing RCA ribbon mics after the company's microphone division closed. In 1998, as the sole manufacturer of parts for the RCA 44, AEA decided to release their own version of the iconic mic, the R44C. They have since continued to advance ribbon technology, and their current range offers a choice of three active designs. The A440 (www.soundonsound.com/reviews/aea-a440) is a premium version of their original design equipped with a Lehle transformer, and retains the same sonic character with an additional 18dB of output and significantly improved noise performance. The N22 (www.soundonsound.com/reviews/aea-n22) has then been optimised for close-miking, with a limited proximity effect and rugged construction allowing it to be placed closely to loud sources whilst retaining a balanced frequency response. Finally, the KU5A (www.soundonsound.com/reviews/aea-ku5a) is an end-address design which, unusually for a ribbon mic, features a supercardioid polar pattern rather than figure-of-eight.

£ A440 £6499, N22 £1099, KU5A £1299

W https://aearibbonmics.com

Audio-Technica

Audio-Technica AT4080 and AT4081

There are two active ribbon models in Audio-Technica's extensive range of microphones: the AT4080 and AT4081 (www.soundonsound.com/reviews/audio-technica-at4080-at4081). Both aim to deliver the smooth, warm character associated with ribbons, but with a more mechanically robust construction than vintage alternatives. The former is the larger of the two models, with a low-end response extending slightly further than its smaller sibling, which features a more compact, 'stick' profile that helps to maximise placement options. The AT4080 comes supplied »

with a custom shockmount, whilst the AT4081 ships with an isolation stand clamp.

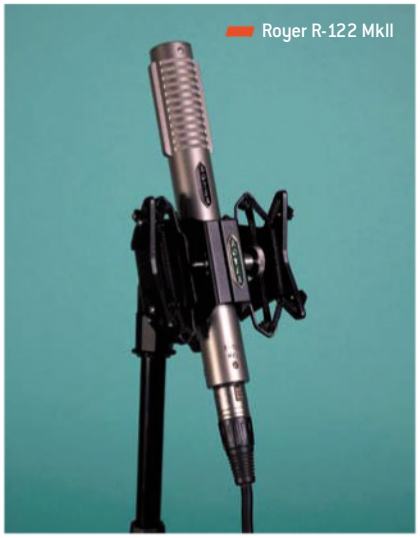
£ AT4080 £849, AT4081 £639
W www.audio-technica.com

Rode

Rode NTR

Rode's only ribbon, the NTR (www.soundonsound.com/reviews/rode-ntr), promises extremely high-end performance at an affordable price. A laser-cut 1.8-micron-thick ribbon allows it to offer an extended high-frequency response, with the cutting process used during its manufacture said to result in smoother edges and increased longevity. The microphone's design minimises acoustic interactions between the body and ribbon, and the frame surrounding the ribbon has been placed in the null of the polar pattern to reduce resonances. A sturdy internal shockmount design also means that the NTR can be mounted directly on a stand without an additional shockmount.

£ NTR £689
W rode.com


Royer R-122 MkII

Royer Labs

Founded by David Royer in 1998 with the aim of delivering new and unique ribbon designs, Royer Labs are often credited as the company behind the modern ribbon mic resurgence. In 2002, they released the world's first phantom-powered ribbon mic, the R-122. The current R-122 MkII (www.soundonsound.com/reviews/royer-r122-mkii) builds on its success, adding a -15dB pad to help tackle loud sources, and a 100Hz, 6dB/octave high-pass filter to counteract bass build-up caused by the proximity effect when close-miking. As with all of the company's R-series designs, the mic features Royer's offset ribbon technology, which helps to maintain a full frequency response during high SPL recordings by placing the ribbon closer to the front side of the microphone.

£ R-122 MkII £2025
W royerlabs.com

Ocean Way RM1-B

Ocean Way Audio

Ocean Way Audio are renowned for their no-compromise approach to equipment design, and their RM1-B (www.soundonsound.com/reviews/ocean-way-rm1-b) is no exception. It is a large microphone with a striking visual design, weighs a substantial 3.4kg, and is said to feature the strongest magnetic field present in any ribbon microphone. Its construction promises to deliver resonance-free acoustical performance, with all-discrete output circuitry offering a clean and natural sound along with excellent noise performance. It is recommended for use on a wide variety of sources, although the company advise against close placement on instruments such as kick drums and bass amps to avoid damaging the ribbon element.

£ RM1-B $3250
W oceanwayaudio.com

Cloud Microphones

Cloud's ribbon mics were designed in collaboration with Stephen Sank, whose father, Jon R Sank, was involved in designing many of RCA's iconic models. As its name suggests, the

Cloud 44-A

Cloud 44 represents the pair's take on RCA's legendary 44-series designs. The Cloud 44-A (www.soundonsound.com/reviews/cloud-44) takes things a step further, employing a built-in version of the discrete Class-A preamp circuitry used in the company's popular range of Cloudlifter devices. Being inspired by the early RCA designs, the mic offers a traditional ribbon sound, and features switchable Voice and Music tone options, with the former cutting low end for a more natural sound when close-miking vocals.

£ 44-A £1699
W www.cloudmicrophones.com

Pinnacle Microphones

The Pinnacle Microphones (formerly Cascade) range is predominantly made up of passive models, but there is an interesting exception in the form of the Fat Top II Active/Passive, which

Pinnacle Fat Top II Active/Passive

on tour with

JAMES "JIMMY T." MESLIN

PUSHING FADERS FOR **DREAM THEATER & JOHN PETRUCCI**

Q A

Tell us about your career and how you started in audio.

I'm James Meslin, often referred to as "Jimmy T." I am a recording and mixing engineer for both studio and live sound. I started my professional career in the studio as an intern. I'd bounce between that and my own home studio, keeping myself completely occupied with record production at all times. As I gained experience, I moved up the ladder in the studio - from intern to assistant and eventually head engineer. In 2015, I worked on **Dream Theater's** double album, **The Astonishing**, which led to an offer to tour with them as a playback engineer. Since then, I have balanced a life of live sound (mixing monitors, front of house, playback, and live show recordings) with my roots in studio work and production.

What's a challenge you face as a studio and touring engineer?

Work-life balance continues to be a challenge. Projects may hit a lull, you'll pivot to new jobs, and then the call comes in that the original project is on a fast track to finish. I've always taken my work home with me, answered the 1 AM phone call to discuss mix revisions, or left the house for a seven-week tour. It's even more challenging when deadlines coincide with being out on the road. Prioritizing time off is healthy and often benefits your work when you're back at it. Finding balance in an industry that ebbs and flows is difficult, but I feel it's important to carve out time to reset.

What is your favorite sE gear, and how do you use it?

I really enjoy a lot of what sE has to offer. Their microphone selection, the **sE Munro Egg 150** monitors (discontinued but worth it if you can find them used), and the various Reflexion Filters are all stellar and make frequent appearances on my sessions & gigs. Years ago, my first big investment in a microphone was the **Gemini II**. It became my go-to vocal mic, and I continue to put it up as an option for any session I do. Despite the great range of sE products, my current favorite microphones are the **RN17** and **VR2**. While very different in what they do, their ideologies are similar. Both models have a unique design, and their sound characteristics give you double the value. The **RN17** is a pencil condenser with a large transformer in line. This changes everything you'd expect from a SDC. The depth, warmth, and fullness it delivers is unmatched, yet the precision of the small-diaphragm mic is not lost. The **VR2** is an active ribbon microphone with all of the woof and body you'd want out of a ribbon microphone partnered with the extended top-end you'd find in a condenser. This makes it the holy grail for guitar cabinets, in my opinion. Miking guitar is often done with multiple mics that are blended to taste. The VR2 gives you a single mic option to achieve that effect. I'm currently on the road with **Dream Theater,** and the VR2 mounted in the **guitaRF** is the only microphone on **John Petrucci's** guitar cabinets. Every night, the VR2 delivers us a consistent result that makes us all smile over the tone. In addition to the VR2, we are currently using the **V7 MC1** capsule on **James LaBrie's** wireless handheld, a **V7** on John Petrucci's vocal, and a **V BEAT** on **Mike Mangini's** snare drum.

Any tracking tips or industry advice?

Always experiment and be creative. Use recording and production as an extension of songwriting, don't be afraid to commit front-end processing, and frequently hit save. For industry advice, anticipate everything to go wrong, have solutions, and keep a good attitude at all times.

"sE delivers products that are reliable, durable, well priced, and above all - sound great. They have proven their equipment can handle traveling the world on the main stage while still offering a product that holds its own in a pristine studio setting."

James "Jimmy T." Meslin, Dream Theater & John Petrucci

read more at **seelectronics.com**

» features a switch that allows it to operate as either a passive or active mic. With the active circuitry engaged, the ribbon is kept consistently loaded, and up to 20dB of extra gain is delivered to the output. The mic is also equipped with a Lundahl LL2913 transformer — which is an optional upgrade on the company's other offerings — that is said to offer an extended frequency range and improved definition.

£ Fat Top II Active/Passive €699
W pinnaclemicrophones.com

HUM Audio ARM-1L and ARM-1S

HUM Audio Devices

HUM Audio Devices' ARM-1L and ARM-1S (www.soundonsound.com/reviews/hum-audio-arm-1) appear almost identical to look at, and at a glance are distinguishable only by the colour of the lettering around their bodies. The difference is more significant that it seems, though: the letters in their name denote the use of two-inch (Long) and one-inch (Short) ribbon elements. As a result, the former is said to offer a more contoured sound which imparts a little more character, whilst the latter is smoother and more tonally balanced. Both employ discrete active circuitry, and feature a switchable 50/100 Hz high-pass filter.

£ ARM-1L & ARM-1R £1905.60
W www.hum-audio.com

Extinct Audio

Extinct Audio's BM9A (www.soundonsound.com/reviews/extinct-audio-bm9a) has been designed to offer the exact sound of their BM9 Viking, but with an active output stage that provides an additional 22dB of gain.

Extinct Audio BM9A

Visually, it draws inspiration from vintage Bang & Olufsen designs, but its modern circuitry and manufacturing techniques offer much improved noise performance. The high-frequency response is said to be as smooth and detailed as you'd expect from a high-end ribbon, with a rich bottom end that is particularly evident when close-miking thanks to a strong proximity effect.

£ BM9A £999
W www.extinctaudio.co.uk

Peluso Microphone Lab

Peluso's TR 14 differs slightly in that, being a valve mic, its output circuitry draws power from a dedicated PSU rather than phantom power. The use of a valve also imparts some additional harmonics, although the effect is subtle — Peluso say that the TR 14 delivers natural-sounding results that respond well to EQ. The rear side of the mic has slightly more low-end roll-off due to the ribbon being placed slightly forward of the magnetic centre, a useful feature that helps to counter proximity effect when close-miking, if required.

Peluso TR 14

£ TR 14 £999
W pelusomicrophonelab.com

ART

ART's AR5 has been designed to offer the sonic qualities associated with classic ribbon designs, whilst its active output stage ensures a consistent performance regardless of the destination input. It offers an extended frequency response when compared to many older designs, and will withstand a maximum SPL of 150dB. The mic comes supplied with a shockmount that promises to virtually eliminate handling noise and vibrations as well as making it easier to position, and plenty of protection is offered by a rugged body construction and steel mesh grille.

ART AR5

£ AR5 $159.99
W artproaudio.com

Samar

Samar say that their VL37A combines a vintage vibe with a modern sound, and offers a versatile solution that works well on everything from classical recording sessions to guitar amps and drum overheads. Its motor structure is based on their acclaimed MF65 model (also used in the passive version: www.soundonsound.com/reviews/samar-vl37), and every model is meticulously built by hand and fine-tuned to ensure consistency between each unit. It boasts a ruler-flat frequency response that extends beyond 25kHz, and is only 0.43dB down at 20Hz!

Samar VL37A

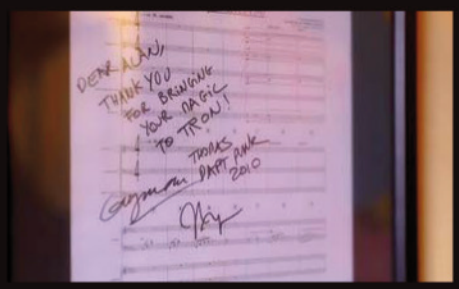
£ VL37A £1273.20
W samaraudiodesign.com

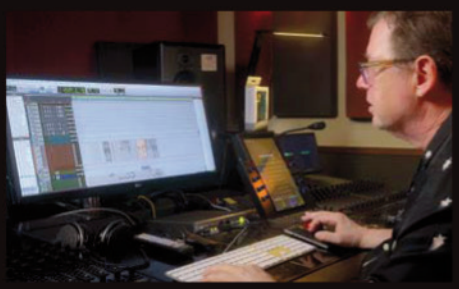
Mesanovic

The Mesanovic Model 2A offers the same extended frequency response and sonic character as the company's passive Model 2 version (www.soundonsound.com/reviews/mesanovic-model-2), a mic praised for its detailed and natural sound. Where the active version differs from many is that the active electronics do not provide any additional gain: they are used purely as an impedance buffer. Combined with the mic's high-turn-ratio transformer, however, the circuitry results in an additional 13dB of output gain compared the passive version, along with extremely low self-noise.

Mesanovic Model 2A

£ Model 2A £1165.83
W www.mesanovicmicrophones.com





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Alan Meyerson: MIXING THE MOVIES

Pop producer and engineer Alan Meyerson was down and out when a chance meeting with Hans Zimmer took his career in a new direction. Thirty years and hundreds of film soundtracks later, Meyerson is Hollywood's first-call score mixer.

We visited Alan in his state-of-the-art Atmos mix room at Zimmer's Remote Control facility to hear his extraordinary life story and find out what's involved in mixing a blockbuster movie score. Then, in our special bonus feature, Alan takes us on a deep dive into his mix of Mark Mothersbaugh's soundtrack to the Marvel classic *Thor: Ragnarok*.

www.youtube.com/soundonsoundvideo

SAMPLE LIBRARIES

UJAM
Beatmaker Cozy
Plug-in Instrument

★★★★★

The latest in UJAM's Beatmaker collection is Cozy, released with the aim of tapping into the lo-fi market that has become so popular in recent years, by offering a collection of lazy, slightly delayed retro hip-hop beats. According to the UJAM website, top-quality samples have been used here and then degraded to achieve the retro feel.

As with all Beatmaker plug-ins, you can select your kit on the top left (10 to choose from with 16 instruments each). To the left of the kit selector you can balance between kick and snare or dial each of them out altogether. Along with the standard drum sounds, you have claps, percussion and accents. The control to the right is referred to by UJAM as their Special control, with each UJAM plug-in having a different feature here. In this instance it's called Nostalgia and allows you to increase the lo-fi feel by adding a little vinyl noise, smoothing out the transients and bringing in some tube warmth, all in one dial.

In the upper-right corner you have an integrated Finisher, which allows you to add 12 different retro effects, with an Amount control for bringing in the level of the effect and a Beat Intensity control for increasing overall velocity.

C1-Eb2 on your keyboard will trigger one-shot samples, while C3-Bb4 triggers various patterns that cover Intros, Verses, Choruses, Fills, Breaks and Endings. With 20 drum patterns to choose from, each containing 23 variations, that gives you a lot of inspiration to get started with. The newer UJAM plug-ins now feature MIDI drag and drop, so you can use the patterns here to get started quickly, then easily tailor the parts to add a more human touch to your finished track. The micro-timing control can also be used to humanise a track by offering a more relaxed quantisation.

Each of the kits can be mixed and matched — if a kit is mostly working for you but the snare is too harsh, switch it out for another by highlighting the drum on the instrument panel and using the controls below to select a replacement. Also in this panel are individual volume, decay, pitch and filter controls for more fine-tuning.

Finally, the master panel in the bottom-right section offers a Filter sweep (ideal for lo-fi), Saturation, a Maximiser and an Ambience control, with the option of selecting your reverb from the menu above. Beatmaker sounds can be sent to individual outputs, thus allowing for even more control over panning, mixing and additional effects.

Cozy is a simple-to-use but very effective plug-in, with a usable collection of drum sounds that are a great alternative to loops and perfectly suited to lo-fi, hip-hop, trip-hop and smooth grooves. *Atheen Spencer*

£60.83

www.ujam.com

Cinematique Instruments
Rytmik II
Kontakt Instrument

★★★★★

SOS reviewed the original Rytmik library from Cinematique Instruments in the January 2021 issue. As with many of the company's titles, it was perhaps aimed primarily at working media composers and had a price tag that reflected that. However, the slightly left-field collection of drum and percussion sounds and cool/creative UI provided a winning — if somewhat quirky — combination. Unsurprisingly, Rytmik II follows the same broad concept and, thanks to an expanded sample base and some thoughtful evolution of the UI, promises to make something that was already very good even better.

Rytmik II delivers some 350 individual sound sources that span sound-designed booms, drum sounds (both conventional and unconventional) and a wide range of percussion-style sounds. Within an individual preset, a combination of these can be organised into the 14-lane step-sequencing environment, with sequences up to 32 steps in length that also offers per-lane sequencing of effects. There are mixing control options for each lane, global effects,

and flexible sound modulation options. All of these functions are spread across three main pages (accessed via tab icons located towards the top-left of the UI) while a fourth page — the default on opening the instrument within Kontakt — provides a very neat browser with tag-based categories to filter your selection from the 500+ supplied presets. Filters such as Organic, Electro, Cinematic, Dark, Ticking, Busy or Lazy hint at the intended styles but, almost wherever you look, there is something interesting, intriguing and inspiring to find. The presets do a great job of showcasing just how good Rytmik II can sound.

There are a lot of very cool features built into the sequencing and effects engines. For example, within the Mixer screen, you can select a sound to access more detailed settings for that sound. This includes its step-sequencing options but also randomisation options that allow you to control which element (pattern, sound, EQ, etc, or everything) get randomised. It's a great way to experiment with changing up a pattern to either give it a different flavour or to add variety. Also excellent are the global attack and decay controls that operate on all sounds within the current preset and allow you to shift it from sharp and snappy to soft and boomy and all flavours in between. Even better, you can automate these and create some fabulous variations in dynamics. And there is plenty more to discover once you dig in.

Rytmik II is the kind of thing Cinematique Instruments do so well. While the price might put it beyond some, working media composers or more experimental pop music producers will find plenty to love here. The UI is cool and encourages experimentation, the sounds are wonderfully quirky, and the extensive preset collection will inspire

straight out of the box. The original Rytmik was very good indeed; Rytmik II is even better. *John Walden*

€196.35

www.cinematique-instruments.com

Sample Logic
Drum Fury Motion

Kontakt Instrument

★★★★★

Sample Logic's original Drum Fury library got a five-star review in the June 2019 issue of *SOS*. Their latest release — Drum Fury Motion — builds on the same broad area of cinematic drums but, this time, combines a sample-based drum and percussion library with the powerful step-based pattern sequencing engine found within some of their other Kontakt-based instruments such as Symphonic AI and Motion Keys. In Drum Fury Motion, a full preset can therefore be built from two of the underlying drum sounds and each of these has its own step-sequencing engine slot, with options to sequence a whole range of note and effect parameters. The product includes over 6GB of sample data, over 1800 instruments and presets and requires the full version of Kontakt (v6.7.1 or higher).

Drum Fury Motion's sounds are derived from many of the same sorts of cinematic-friendly drum types as the earlier library but, compared side-by-side, the new release also includes some more contemporary sources such as beatbox, metals and synth-based sounds. The other obvious difference between the two products is in triggering the sounds. Drum Fury is designed to let you play the sounds via your MIDI keyboard (a more conventional sample library approach) while, in Drum Fury Motion, you are triggering the step-sequencer patterns. It's a different way of working but both approaches have their merits.

The step-sequencing environment is pretty deep and does require a little time to fully appreciate. It's time well spent though, as this is powerful stuff. It's also where that

modern edge to the sound is enhanced further because the various effects options — both within the step sequencer and in the Mix and Master sections of the UI — make it possible to easily move the sounds between conventional ambience treatments through to fully-blown sound design. It can do conventional drum sounds but if you want the option for high-tech and processed percussion within your score, Drum Fury Motion can also fulfil that need.

The Sample Logic team obviously had a lot of fun with these tools creating the extensive list of presets, and media composers are going to find lots here that can be dropped straight into an action or drama cue. Once you dig in, creating your own presets and rhythms is easy enough but the UI also offers a whole range of randomisation options that quickly let you search for some additional variations or inspiration.

Drum Fury Motion sounds great and, with its pattern-based playback and cutting-edge sound-design options, it would make a great complement and/or alternative to a more conventional 'epic drums and percussion' library (such as the original Drum Fury). It is obviously aimed at (and priced for) busy media composers but if you need tools that both inspire and help you meet a tight deadline, Drum Fury Motion would make an excellent choice. *John Walden*

$299.99

www.samplelogic.com

Spitfire Audio
BBC Symphony Orchestra Piano

Plug-in Instrument

★★★★★

Currently retailing at a cool £156k, the nine-foot Steinway Model D-274 grand has long been the instrument of choice for leading concert pianists, composers and recording artists. While those that can afford the real thing are unlikely to be seeking a digital alternative, the rest of us can take heart from the current abundance of high-quality Steinway D-274 sample libraries. The latest to cross my desk is Spitfire's BBCSO Piano, recorded in Maida Vale Studios using the same microphone positions as the main BBCSO library (reviewed in *SOS* November 2019). I'm pleased to say Maida Vale Studios has now been purchased by Hans Zimmer, who we assume will preserve the building's recording studio heritage.

BBCSO Piano is available in Professional (35GB) and Core (2.8GB) versions, the

former boasting the full complement of 15 mic signals and the latter offering only a single 'Mix 1' signal. In all other respects the two products are identical. A smaller version of the piano is also contained in the free BBCSO Discover library, but it's not separately available. The samples are presented exclusively in Spitfire's dedicated plug-in, which runs in your DAW or VST host.

Sampled at six dynamics with four round robins and dedicated pedal up and down performances, the instrument incorporates volume controls for hammer release and pedalling noises — these small sounds undoubtedly add realism, but recording engineers will be glad to be able to turn them off. You can widen or narrow the stereo image of the close mics, and choose between a full 15-channel signal mixer and a simplified close-far mic position slider.

For this project Spitfire hired BBCSO full-time pianist Elizabeth Burley, who is intimately acquainted with the piano and the room it occupies. The relationship between player, instrument and acoustic space has produced a beautifully balanced set of samples with imperceptible transitions between dynamic layers — I was surprised to learn only six were recorded, it feels like considerably more. The piano has an attractively bright, open sound which softens to an intimate murmur when played quietly and scales up to a thunderous crashing when you give it some welly.

This Steinway D supplies a traditional classical sound straight out of the box — the default Mix 1 signal has presence, depth and enough room sound to blend perfectly with the instruments in the full BBCSO library. For a more immediate pop sound, load the close mics and add ambient mics or extra reverb to taste. It goes without saying that Paul Thompson is "really excited" about this instrument (he says that in all his walkthrough videos). However, this time the Spitfire co-founder's personal enthusiasm for the instrument is evident. Having spent some time playing this fine sampled piano, I'm beginning to feel the same way. *Dave Stewart*

Professional £149, Core £79

www.spitfireaudio.com

Audio examples of this month's libraries are available at www.soundonsound.com.

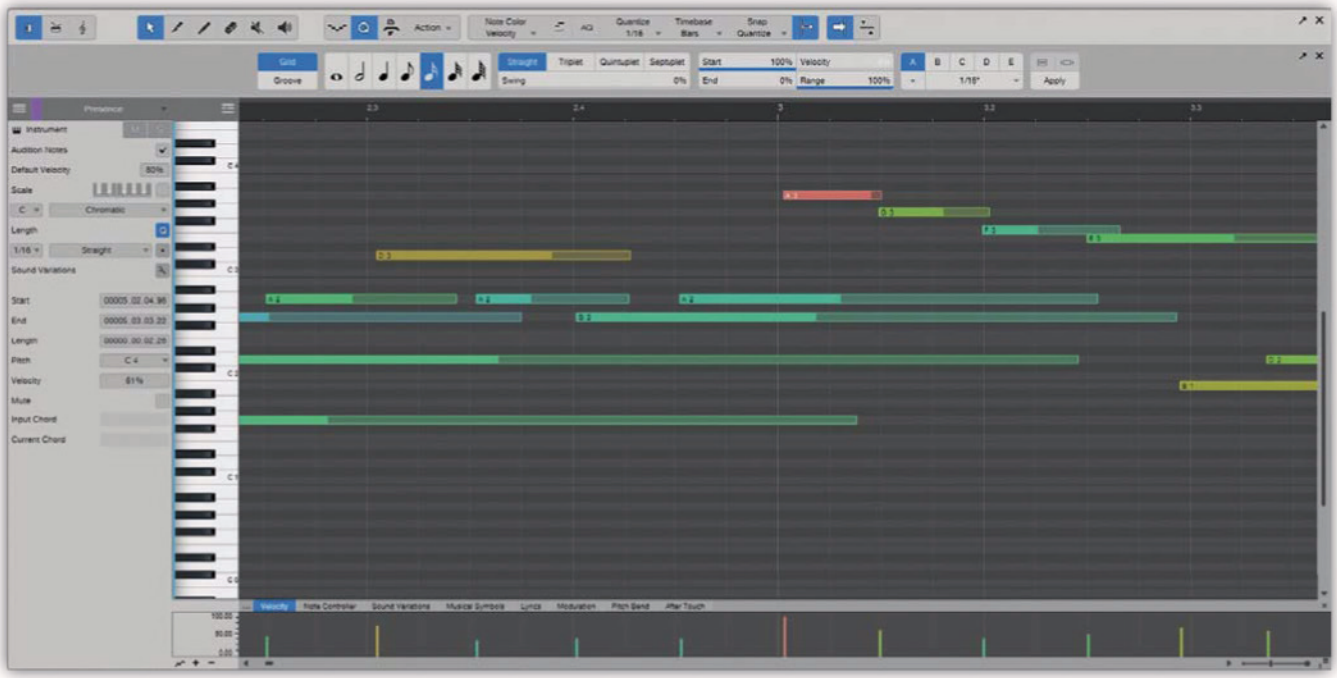
Clean up your MIDI with Studio One's comprehensive quantise options.

ROBIN VINCENT

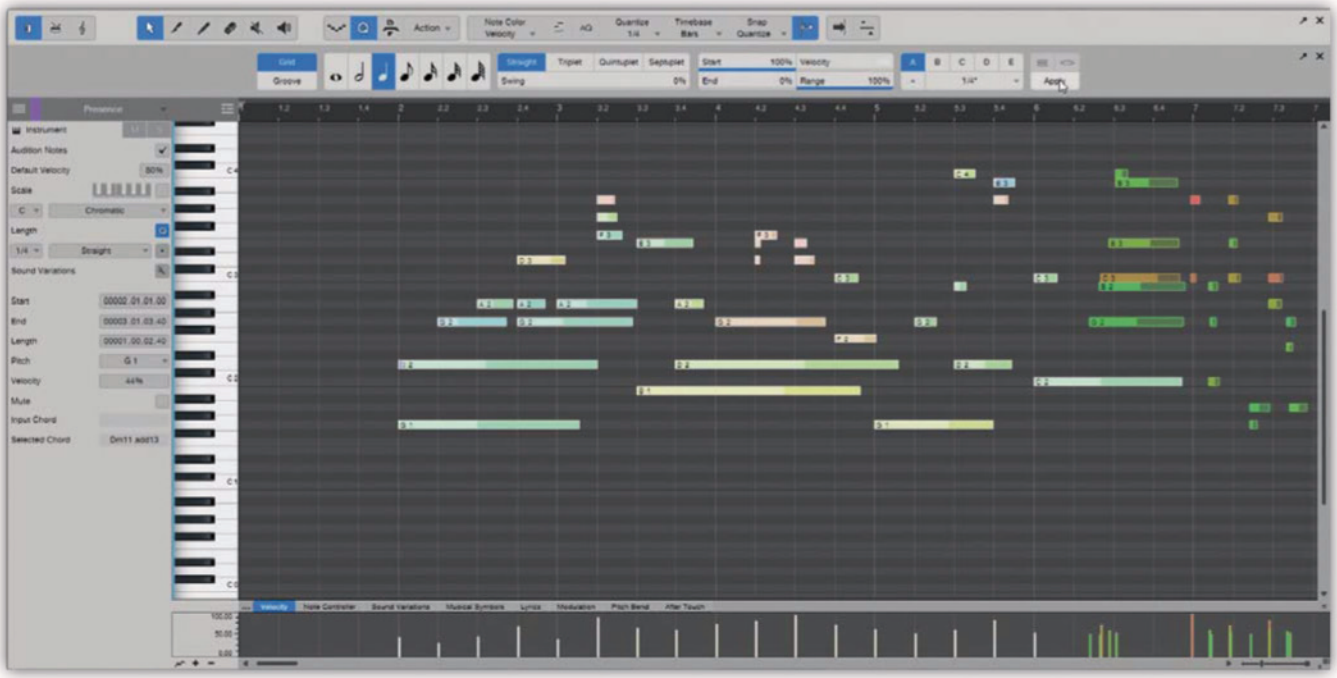
Last month we swam in the murky waters of Snap and discovered the intertwined sunken treasures of Grid, Timebase and Quantize. This month we're going to dive deep into the quantisation engine and see if we can find out how to manipulate time for the better.

To get fully engaged with quantisation, we need to open the Quantize Bar. You'll find it under View / Additional Views / Quantize. You can pop this bar out into its own floating panel by clicking the diagonally up-and-right-pointing arrow next to the X on the right. Interestingly, there are separate Quantize Bars and floating panels for the Arranger and the Editor, although there's plenty of overlap because PreSonus like you to be able to do things from at least three different perspectives.

It will probably make more sense to explore quantising in the Edit window, so let's open the bar via a different route. Create an instrument track, double-click in the track to generate a clip, double-click the clip to open the Edit window and press the big



An unquantised MIDI part. Note how the note starts don't quite land on the grid.



The same part, but quantised (and zoomed out). The note starts have been forced to the grid, which has been set to quarter notes (the highlighted crotchet icon).

Q button in the Edit window toolbar to reveal the Quantize Bar. Detach it if you wish.

What Is Quantising?

Quantisation is the aligning of data to some sort of predefined grid. In the context of Studio One, quantisation is to do with the timing of note and audio events. In other systems, most notably control-voltage-based modular synthesizers, it can also be about pitch, but that's not the case here.

In its most basic form, quantisation will correct your sloppy playing so that all the notes will fire at the right time according to the specified tempo and time divisions. It can both perfect and ruin your performance. But ultimately, all things being equal, quantisation is immensely useful in making notes happen at the right time without you needing to be a virtuoso.

Studio One's quantisation engine has a lot going on, so let's get into it.

The Q Continuum

Let's begin with some notes. Load up your favourite virtual instrument on an instrument track. Enable the metronome on the transport bar. I'm assuming you have a MIDI controller keyboard attached. Hit Record and play something along to the metronome for a few bars. We can now see our notes captured on the grid of the piano roll editor.

If you zoom into the notes you'll probably notice, if your playing is anything like mine, that they don't fall perfectly on the grid lines. Now you could go in with your mouse and adjust the notes individually so that the timing is perfect. However, this is exactly what quantisation is for. Select any rogue note, hit the Q key on your keyboard, and it will pull itself in line with the grid based around your quantise settings.

The Quantize Bar

Looking at the Quantize Bar, you can see that we have options: some note value choices, some rhythmic shenanigans, some parameters, some presets and an Apply button.

We have a choice between Grid and Groove; we'll stick with Grid for the moment, as this is most common form of quantisation. You can see the grid on the piano-roll editor. As you click on the note values next to the Grid button, you will see the grid lines change to reflect the value of the selected note. Now, these note values don't have anything to do with the length of the notes you played. Instead, they dictate the resolution of the grid to which the start of your note will be quantised. So, with a semibreve (whole note), the quantisation is a whole 4/4 bar. For a minim or half-note, you get two grid lines in a bar. With a crotchet or quarter-note, you get four grids lines in a bar and so on. But again, your notes are not getting turned into crotchets and quavers; this is about defining the grid in the bar that notes will snap to.

The Quantize Bar is where you define how your notes will be moved.

The best way to understand this is to select all your notes and play with the different quantisation note values. Either hit Q or use the Apply button to shift your notes around to the nearest grid line. Use Undo in between to reset your notes to their original off-grid performance.

You may not want all your notes quantised to the same setting; in this case, simply select different notes or bunches of notes and apply the desired setting just to them. In some cases, in order to retain a bit more of your characterful playing style, you might only want to quantise a few of those notes.

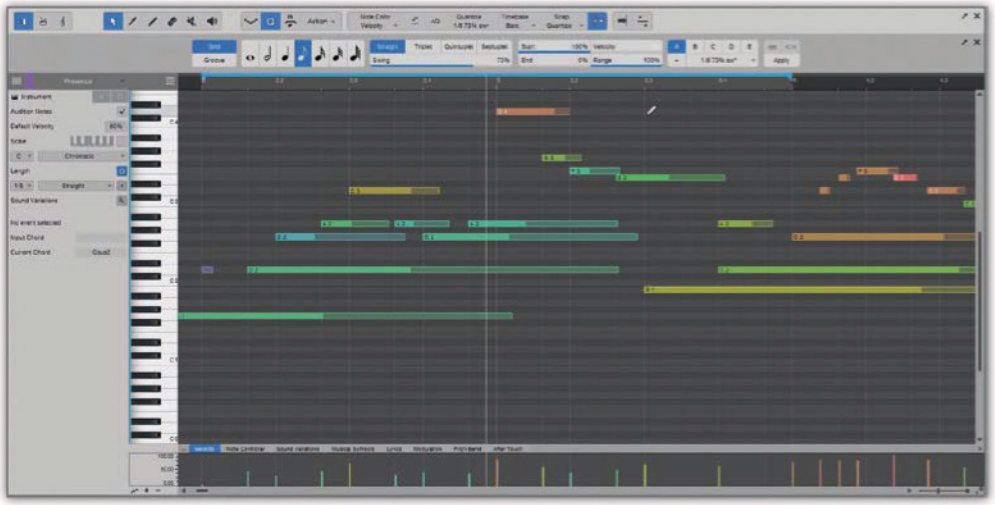
The next section of the Quantize Bar deals with triplets (three notes in the space of two), quintuplets (five notes in the space of four) and septuplets (seven notes in the space of eight), along with a nice and long Swing slider. As before, watch the grid as you select different options to see what your notes could be snapping to. Swing pushes the off-beat grid line forward in time to create a more relaxed feel. It can be a very interesting section to play in to find new rhythms and feels that you might not have originally intended.

For a bit more nuance in our quantisation we can explore the Start, End, Velocity and Range parameters.

Start defaults to 100 percent, meaning that when quantised, every note will be moved to the appropriate grid line. As you pull the value down, the notes are less and less attracted to the grid. So rather than forcing your performance onto perfect lines, you can gently nudge them towards the correct timing. With a low Start setting, you can use multiple taps of Q to encourage them closer and closer to the grid until you get the feel you're after.

End defaults to zero percent and deals with the quantisation of the length of the notes. Again, the quantise note values are not dictating the exact length of the note, rather that the end of the note is being quantised to the nearest grid line.

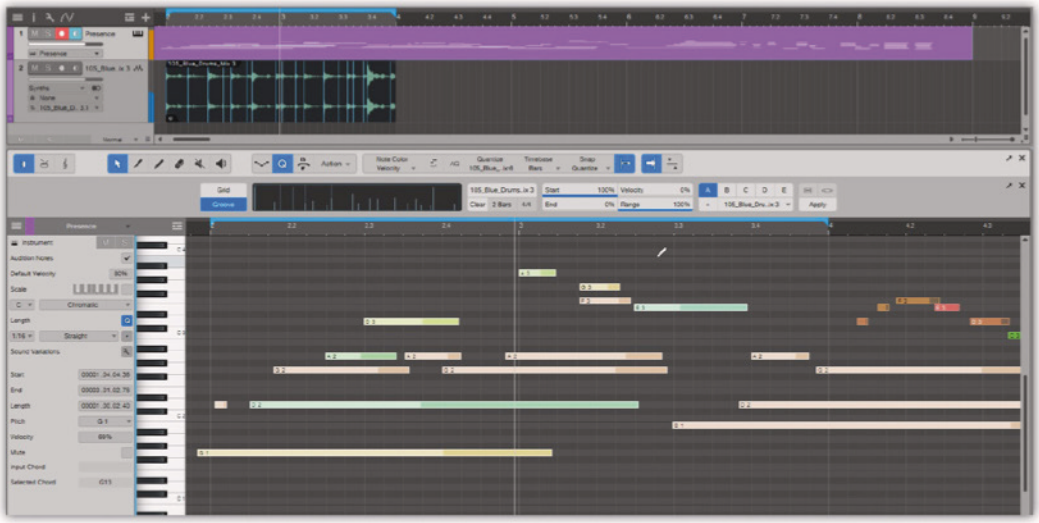
The **Quantize Range** is slightly odd. At 100 percent, you can be assured that all the notes in the selection will be quantised. »



The Swing slider pushes off-beat grid lines forward in time, giving you a loping feel.

» At lower Range settings, some of the notes are not quantised. As far as I can make out, the Range is about the distance of the notes from the grid. Only the notes within that percentage distance of the grid line get quantised as the percentage comes down, while the others are ignored. The manual suggests trying different settings and seeing what happens.

Velocity isn't available in Grid mode, and we'll come onto Groove mode in a moment.



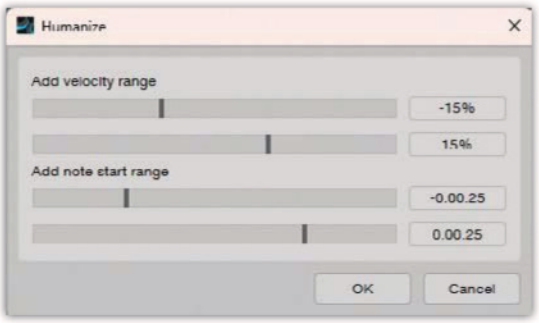
Here, an audio drum loop (top) has been used to create a Groove, to which the MIDI notes below have been made to conform.

You may have noticed that the section with A B C D E across it changes as you select different settings. This is an invaluable quantise preset panel. Once you've set the quantising how you like it, you can hit the '+' button and add the settings as a preset. Each of the lettered options can have a preset allocated, so you can easily switch between them.

Humanize

Before we get all loose and groovy in the Groove mode, let's look at a Humanize. Humanize is quantisation's nemesis. It comes from the idea that quantisation can suck the humanity out of a performance and so when we Humanize, we reinject our flaws and sloppiness.

You would expect Humanize to be part of the Quantize Bar, but actually it's hidden away under the Action menu. There are a few other useful Quantize actions in this menu. You can apply a 50 percent quantisation, which is like setting Start to 50 percent. You can quantise just the ends of notes. Restore Timing will essentially remove the quantisation and return your notes to their original position. Freeze Quantize is when you want to make the quantisation permanent, perhaps because you want to quantise again from your already quantised position. Be aware that this replaces the original notes with the quantised ones, and removes the option to Restore Timing.



The Humanize menu does the opposite of quantising: it injects randomness to the timing and velocity of your MIDI notes.

Meanwhile, back in the Humanize panel, the idea is to push notes off the grid in either direction, and to give some variation to the velocity. The movement is slight and is modelled on the way most humans play. This can be very useful for breathing life back into heavily quantised notes. It's also a great thing to use if you've converted a pattern into notes, or have placed them in step mode or by hand. It's subtle and believable.

Groove Mode

If you hit the Groove button beneath the Grid button, all the quantisation values get replaced with a curious blank space showing a few even grid lines. This has the remarkable ability to extract grooves from audio recordings or instrument parts and then let you apply them to your MIDI notes or audio. A perfect example would be that you have a sampled drum loop and you would like all your other instruments to play along to the feel of that loop.

All you have to do is drag your event into the blank Groove space. This falls over itself a bit where when you drag an audio file; the editor swaps to the audio editor, so you can't drop it into the Quantize Bar you had set up for your MIDI notes. But don't worry, just open the Quantize Bar on the audio editor and save the Groove as a preset. Then you can open it in the piano-roll editor.

To pull a Groove off an audio file, Studio One has to do some clever transient detection. You have a lot of control over how that works and can apply all sorts of tweaking to get the best possible Groove. But we'll cover that in another workshop. For now, just know that if you like the feel of an audio loop you can pull that directly into the quantise engine and everything will pull towards that feel. You can do the same with a bar or two of MIDI notes, and in fact, you can force audio tracks to follow the MIDI performance in a very similar way. We will get to that another time.

The Velocity parameter now comes into play: along with the note positions or transients, you can also use the velocity or peak data of those events and project that same character onto the velocity of the selected notes. This whole Groove thing is remarkably good.

Deja Q

One last thing to mention is Input Quantize, which can save you a whole load of time by making things right as you are recording your performance. It's enabled using the IQ button, which should be on the toolbar alongside the other quantise settings. If you don't see it, you can make it appear by customising the toolbar. Once set, any playing from your MIDI keyboard will be automatically adjusted to conform to the quantise settings.

SOUND ON SOUND

PODCASTS

Gain Staging - Episode 3

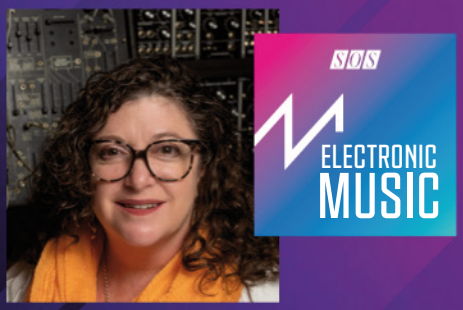
Preparation for your mix

In the third episode David continues looking at preamps, outlines the use of VU Meters and talks about the benefits of balancing a track prior to mixing.

Clark - Sus Dog

Recording his new album

Chris Clark, known professionally as just Clark, reveals how he got signed to Warp Records, his recording and production workflow, and how Thom Yorke ended up as a guest vocalist on his latest album, Sus Dog.



ARP Synth Legacy

The Alan R. Pearlman Foundation

Dina Pearlman talks about her father Alan R. Pearlman who founded Tonus in 1969, a company that would soon become ARP Instruments, and began producing synthesizers such as the legendary ARP 2500, 2600 and Odyssey.

EastWest Composer Cloud+

Behind The Brand

Doug Rogers is President of EastWest, the Soundsonline website and owner of EastWest Studios. Here he chats to Sam Inglis about what it takes to develop their award-winning sample library releases and how you can access everything via their subscription service, Composer Cloud+.

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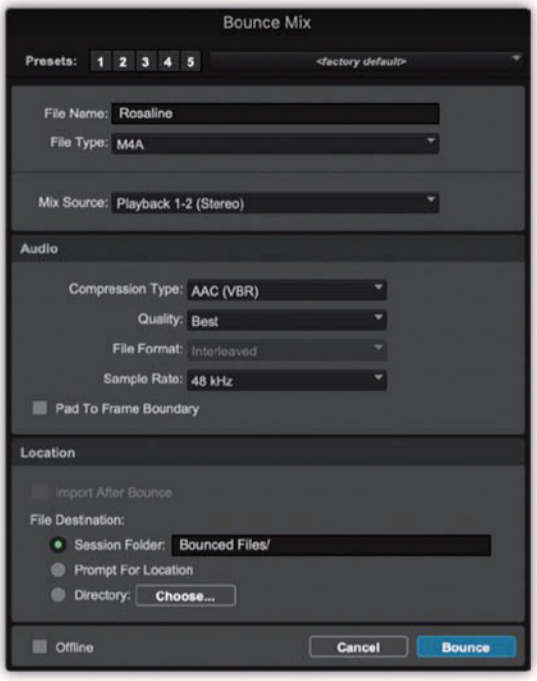
Bouncing a mix isn't the only way to get your audio out of Pro Tools.

JULIAN RODGERS

After all the work has been done and the mix is finally sounding the way you hoped it would, what comes next for the engineer working in Pro Tools? Usually it's going to go as follows: Bounce Mix, type in a meaningful name (which hopefully doesn't contain the over-optimistic word 'final'), and go and make a cup of tea, secure in the knowledge of a job well done. But is that really all there is to it? In this month's article I look a little deeper at what happens between finishing the mix and finishing the job.

Faster, Better?

It wasn't all that long ago that Pro Tools bounces were strictly real-time affairs. It was only 10 years ago that bouncing a mix meant sitting through the whole mix in real time. (Of course, when real-time bounces were first introduced, the fact that it was real time as opposed to slower than real time was the selling point!) However, computers inevitably get faster, and by the time Pro Tools 11 finally introduced faster-than-real-time bounces, real-time bounces seemed positively archaic.



The Bounce Mix dialogue. Since Pro Tools 2023.3, you now have the option, on Macs, to bounce your mix in M4A format.

If you regularly export different versions of mixes, it can be useful to send your source tracks to multiple outputs for bouncing simultaneously. To do this, hold down Control (on a Mac) or Start (on a PC) to select additional outputs.

The shortcut Command+full stop to abort a bounce is prominently displayed in the Bounce Mix window, and for good reason. How many times have you been bouncing a mix in real time and realised, under the scrutiny of the fact that this may well be the definitive version, that you spot a glaring error, one of those "What was I thinking?" moments? Years ago I used to teach Pro Tools and when gathering in projects burnt to CD (yes it really was that long ago) I used to ask "So, who's given me the blank one? One of you will have..."

That anyone would hand over a mix to a client (or a tutor) without checking it still surprises me, and this is why real-time bounces still have a place alongside offline bounces. However, that isn't to say that I would willingly go back to real-time bounces without the option of offline. A mix has to be listened to, all the way through. And when you get into that supposedly perfect mix and spot the facepalm moment, remember that shortcut Command+full stop. If you use outboard equipment then offline bounces will probably have passed you by. They are strictly an in-the-box workflow.

Bounce Mix Window

If you're on a Mac and running Pro Tools 2023.3 or later, a new export option has been added. M4A complements the venerable but still popular MP3 as a compressed output format. The mix source drop-down allows you to select any software or physical output as the source of your bounced file; if you have Pro Tools Ultimate or Pro Tools Studio you can simultaneously create bounces from multiple sources. This is extremely useful if you regularly create different versions of mixes. If you find you do this often it is very worthwhile setting up a template to accommodate this, and an invaluable trick here is to route a track to multiple outputs simultaneously. While this can be achieved using sends, if all you want to do is route identical levels to multiple destinations then holding Control (on a Mac) or Start (on a PC) whilst assigning a track output allows you to route to more than one output at the same time. A '+' next to the output indicates multiple destinations.

Precisely what gets bounced is straightforward enough but there is

something to keep in mind. By default, Pro Tools bounces audio from the main output as set up in the I/O Window, from the beginning of the timeline to the last clip on the timeline. If you make a timeline selection then only that selection will be bounced, allowing you to make quick bounces of specific sections. My preferred method of bouncing a track is to select the last clip on the timeline, hit Option+Return to create a selection back to the beginning of the session, and then manually extend the end of the selection to accommodate any reverb tails. Adjust the start of the selection if necessary and bounce. All settings can be stored as presets so if you find your bounces regularly use different settings, consider saving them to the five quick presets at the top of the window.

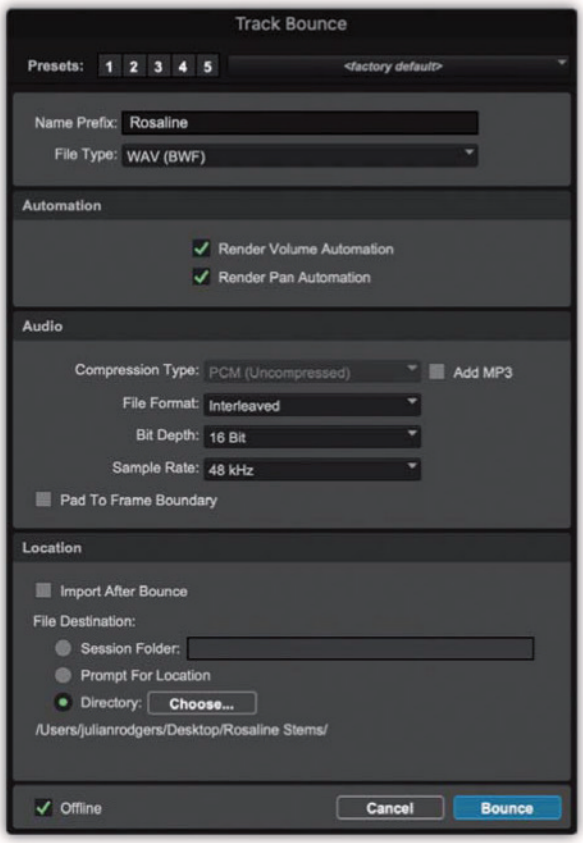
Print Tracks

You might think that outputting your mix is adequately covered by the Bounce to Disk dialogue, but there are lots of times when you might want to take a different approach. In the old days, before Pro Tools 11, the audio engines for Pro Tools TDM systems and native systems were different, and as a result sounded different. There was a popular school of thought in those days that printing mixes in real time to an audio track gave superior results to bouncing to disk. This is no longer the case, but there are still workflow reasons for printing to a track rather than bouncing to disk.

If you are working on a three-minute pop songs and find you need to make an edit and then output your mix again, then bouncing to disk will be fine. However, on longer-form material it can be helpful to print to a print track and then, if edits are necessary, drop in to record across those sections to update the print track. While I've never done this on a music project it's how I regularly do podcast edits, and it saves me a great deal of time.

Bouncing Tracks

The Track Bounce option can be accessed by right-clicking a track name in the Edit or Mix window. It can be used with audio tracks, routing folders and auxiliaries, and is a fast way to create stems. The requirement for stems for remixing collaboration and even mastering is more prevalent than ever and while debates rage about exactly what constitutes a stem, whether you output every track in a project and call those stems or use the term correctly and apply it to groups of related tracks, you can create them quickly using Track Bounce. Simply select the tracks you wish to export individually and either right-click on the track name or use the shortcut Command+Option+Shift+B (Control+Alt+Shift+B on a PC) and set up the bounce window to your requirements. Assigning a destination is particularly important when dealing with a group of related files like stems, and I also find the name prefix field useful at times like this. The option to include volume and pan automation can be used to ensure that when your stems are recombined at unity gain in a DAW your mix will be correctly reproduced.



Track Bounce lets you bounce multiple tracks, routing folders and auxes at a time.

Exporting Clips As Files

Sometimes both bouncing to disk and printing to a track are overkill. For example, while dedicated sample editors exist, I find it easier and faster to work in Pro Tools even if I'm just editing a single drum hit sample or the like. For these purposes plug-in processing is often superfluous, so it isn't necessary to capture the contribution of the mixer and its insert slots. On these occasions exporting clips as files is a much better option. Right-click on a clip in the Clips List, or select the clip you want to export in the Edit window. From the menu that pops up, you can export the clip 'as is', without any contribution from the mixer or inserts. If you select a clip on the timeline it should automatically become highlighted in the Clips List, and *vice versa*. If this doesn't happen you need to check the editing tab of the Pro Tools Preferences, where the 'Clip List selection follows Edit selection' and 'Edit selection follows Clip List selection' options control this mutual selection of clips on the timeline and in the List.

The keyboard shortcut Command+Shift+K (Control+Shift+K on PC) will also open the Export Selected window. This is faster than bouncing as it doesn't necessitate the processing involved in bouncing a mix. I use this frequently in combination with print tracks. This technique can also be used if you are printing a mix to a print track, as in this case processing will already have been captured in the print. However, it's important to consolidate your print track if it contains edits, as individual clips on a print track will be exported as individual files.

So if you ever find yourself soloing tracks one by one and bouncing mixes, keep in mind that there there alternative ways to get audio out of Pro Tools, ways which are sometimes exactly what you need.

"If you make a timeline selection then only that selection will be bounced, allowing you to make quick bounces of specific sections."

JOHN WALDEN

When it comes to MIDI editing, Cubase's MIDI Logical Editor can do magic!

If I had to identify options for a 'most underutilised features in Cubase' list, prime candidates would be the Project Logical Editor (found only in the Pro version) and the MIDI Logical Editor (found in the Pro and Artist versions). Both are incredibly powerful editing tools but, because they use Boolean logic (commands can be chained together to make 'true or false' decisions, which determine the eventual editing outcome), they can also seem somewhat intimidating. It'd be a great shame if you let that put you off, though, because both are incredibly useful and have the potential to streamline lots of repetitive editing tasks. Fortunately, Steinberg include some excellent presets for them in Cubase, and unpicking an example or two provides a fairly gentle entry into these powerful tools. In this column, I'll do just that for the MIDI Logical Editor, which is, as its name suggests, a tool for doing things to your MIDI data.

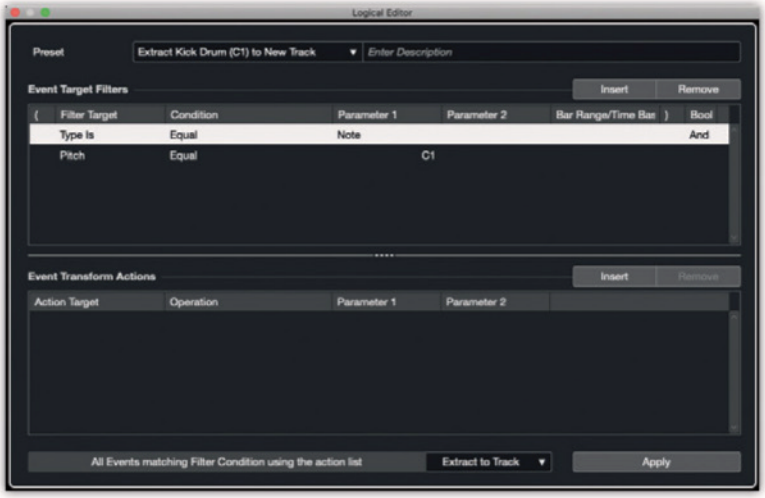
Double Your Kick

Let's start with a preset that's easy to understand, yet turns a very common MIDI editing task into something close to a single click: Extract Kick Drum (C1) to New Track. Layering multiple sounds is common practice with MIDI-based drums so, for this example, let's assume we have created a MIDI clip that triggers our main drum virtual instrument (VI), and we want to double the kick performance with a second VI.

The first screen shows the content of the preset (the preset system is accessed at the very top of the UI). The upper panel (Event Target Filters) lets you specify criteria to define what MIDI events are selected for editing. In this case, there are two criteria used. The first simply states that the Filter Target 'type' is 'equal' to a MIDI 'note', meaning that all MIDI notes (and nothing else) are potential targets for selection. However, the second line refines that selection by specifying that the note's pitch must be equal to C1 (that's the default MIDI note used for the kick drum, though you can change this if desired). The entry in the Bool column (on the right of the UI) is worth paying attention to. In this case, it is set to 'And' (as opposed to 'Or'). This means that the MIDI Logical Editor will only select items that meet both the first and second criteria. For this preset, that means it will only select MIDI notes whose pitch is equal to C1.

In this preset, the GUI's lower panel (Event Transform Actions) is empty. I'll cover this section below but for now note that, with no entries here, the selected MIDI data (notes with a pitch of C1) will not be altered in any way. However, at the very bottom of the GUI, Extract To Track has been specified — so when we hit the Apply button, all the selected C1 notes will be copied from our main drum MIDI clip into a new MIDI clip and placed on a new track. Hey presto! We can now assign this track to an additional kick drum sound, to layer perfectly with our original.

While this is an editing task that you could perform manually, that requires a number of steps. Using the MIDI Logical Editor preset is faster — and faster still if you assign this preset to a key command (the Key Commands window has sections covering both the MIDI and Project Logical Editors). For presets you use regularly, the task then becomes a single click, whether for one selected MIDI clip or several.



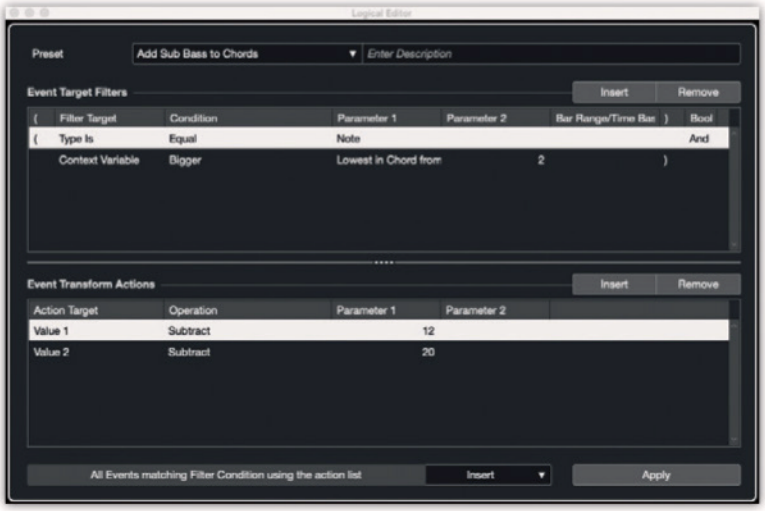
The Extract Kick Drum (C1) to New Track preset.

Add A Bass Note To Chords

Now we're familiar with the MIDI Logical Editor concepts, our second example preset, Add Sub Bass To Chords, can be explained fairly easily.

With a suitable MIDI clip selected, the Filter Target criteria in the preset will search through the clip to find notes. If it finds a time position where at least two notes are playing (a simple definition for a chord), it will select the lowest note in each chord.

The Transform Action panel is also used here, and has two entries. The first subtracts 12 from Value 1 (in this context, Value 1 is note pitch), while the second subtracts 20 from Value 2 (note velocity). Also notice that the specified action is 'Insert'. So, when we click Apply, this preset selects the lowest note in every chord in the clip, lowers its pitch by an octave, reduces its velocity value by 20, and then 'inserts' those transformed versions of the notes into the clip, leaving the original notes intact. So your clip now contains



The Add Sub Bass to Chords preset.

The 10% to Downbeat 4_4 preset (at the top), with the MIDI before (middle) and after (at the bottom) the preset has been applied.

a lower (octave down) bass note for all the identified chords. Ta-da! A simple way to beef up any MIDI part containing chords.

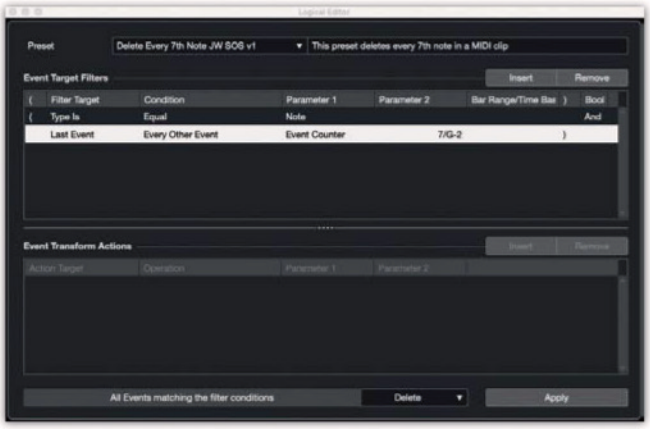
Incidentally, the most common of these (sometimes rather cryptic!) labels (Value 1, Value 2, etc) are explained within the Operation Manual PDF, should you need a reference.

Beat Accenting

Our final preset example demonstrates something a little more sophisticated: the 'note accenting' preset, called +10% to Downbeat 4_4. As in the previous examples, the Target Filter is looking for MIDI notes and, for the notes selected, Value 2 (note velocity) is transformed by multiplying it by 1.1. However, it's the additional Target Filter criteria that make this interesting. These all consider the position of the note within the bar, and each defines a position range around the main four beats of the bar (this is done using PPQ values; Pulses Per Quarter note, which is MIDI's unit of timing).

Notice that the Bool term used for these Position filters is 'Or'. This means that, to be selected for editing, the MIDI item must both be a note 'And' (the Bool term used in the first criteria) an event which sits within the first Position timing range, 'Or' in the second position timing range, 'Or' the third... and so on. Notice also that the last Position criteria is there to catch notes at the very end of the bar; that is, notes played 'early' at the start of the next bar.

The screenshot shows what this preset does to the MIDI velocity data for a simple MIDI clip created using the Arpache SX MIDI plug-in. In the original pattern, every note has the same velocity but, after the preset has been applied (actually, I applied it five times to make the changes more obvious), notes falling on or very close to the beats are emphasised, giving the performance a stronger rhythmic character. You could easily apply this preset to any MIDI clip (for example, MIDI drums), and similar presets are available that emphasise different beat combinations. You can drag within the position displays if you want to adjust the range specified in these criteria. If you like this one, then do check out the Crescendo In Cycle range preset — that's also brilliant.



Want to create some semi-random changes to your MIDI parts? The Event Counter parameter provides one possibility when added to a suitable preset.

Homework

For your MIDI Logical Editor homework, I'll leave you to ponder one final screenshot that shows one of my own DIY presets. As the preset title suggests, for the selected MIDI clip(s), the first Filter Target criteria is looking for MIDI notes. However, in the second criteria, for the Last Event type (the MIDI notes), only Every Other Event will be selected and, in fact, by then using the Event Counter (Parameter 1) set to 7 (Parameter 2), only every seventh note within the sequence ends up selected.

We could then apply a number of different Transformation Actions to process the selected note but, in this case, I've simply chosen to Delete them. Having deleted every seventh note in the original MIDI clip, if we apply the preset a second time, the selected (and then deleted) notes fall in different places. Repeated applications to something like the steady arpeggio pattern shown earlier will break up the regularity of the notes in an almost random fashion, so can create useful variety. You can increase this sense of randomness by creating a second preset that uses a different value (Parameter 2) for the Event Counter. If you create key commands for each preset, they're then easy to execute on the fly — and, if you don't like the results, just execute Undo until you get back where you started. Of course, if doesn't just have to be Delete; you could apply Transform Actions to change the pitch, velocity or even the position (which can be an interesting way to add variations to the timing/feel) of the selected notes.

The Power Of Logic

This short introduction won't make anyone a MIDI Logical Editor ninja overnight, but it does shows how gentle exploration and editing some of the supplied presets can take you towards understanding what's going on under the hood. Hopefully, it also shows you that the MIDI Logical Editor needn't seem so intimidating — and reveals some of the potential of this powerful Cubase feature!

Now that your drums are organised, it's time to whip them into shape!

DAVID RICARD

It's time to wrap up this series on working with live drums in Logic Pro. So far, we've discussed grouping, organising Track Alternatives, track stacking, phase alignment, gain staging, and constructing our final comp. We've also looked at approaches to routing the individual drums to busses and creating sends for processing.

Now it's almost time to finally start time-editing the drums.

Prep Work

We not only want our drums grouped, we also need to ensure that groups are enabled (Shift+G to toggle on/off). You can easily tell if groups are enabled or not; simply click on one of the drum tracks or regions, and if all the other regions become selected, groups are on.

Confirm in the Group Settings that Editing (Selection), Quantize Locked and Track Alternatives are all ticked.

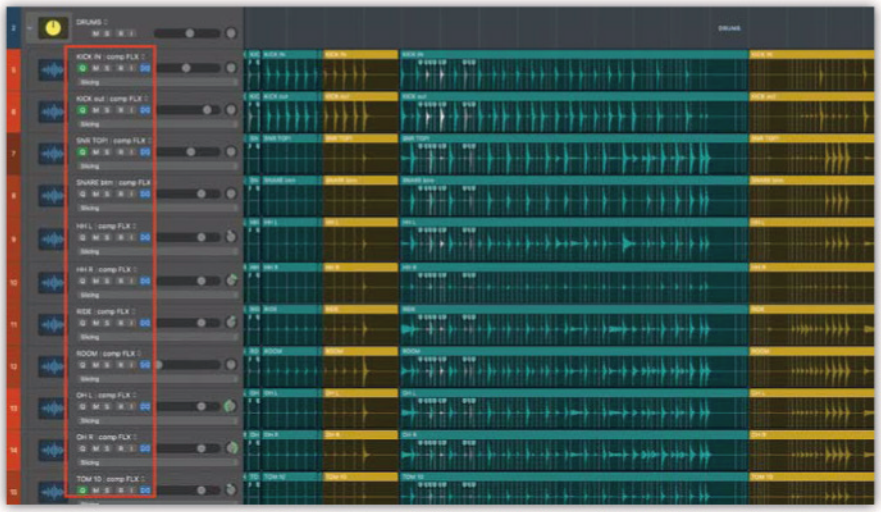
If you have a comp that you're satisfied with, duplicate the track alternative and add 'Flex' to the name. Now, whatever timing edits you make, you can always revert back to the clean comp.

Flex Factor

Besides tuning and timing individual tracks, Flex allows you to time edit multitrack drums. You'll notice a green Q button on each track. This is for quantise reference, ie. telling Logic which tracks to consider when determining how to line things up. Enable this button on your inside kick, snare top and tom tracks. This means that Logic will, for example, use the snare top track as a reference for snare drum timing, rather than the snare in the overheads, room mics and so on.

Enable Flex view (Command+F) and engage Flex on each drum track by clicking on the 'twisty bow tie' icon. If your tracks are grouped (as they'd better be!), clicking it on one track will enable Flex on all of them.

If Logic doesn't pick the Slicing algorithm automatically, choose it from the drop-down menu. While it may be tempting to choose the Monophonic or Rhythmic option, rest assured that Slicing is the one to use for optimal drum editing results. After Logic does a bit of analysis, transient markers will now be visible on each hit of each track. If you see hits that don't have transient markers, there's no need to panic. You can easily add them by clicking on the start of the transient manually.



Here, my drums comp has been duplicated and renamed to indicate it contains Flex edits. Quantize Reference has been enabled on appropriate tracks, and the Slicing algorithm is selected. The different coloured regions indicate that they come from different takes; the highlighted markers are manual edits.

If you zoom in on the tracks, you can then see to what degree they line up to the grid.

Manual Labour

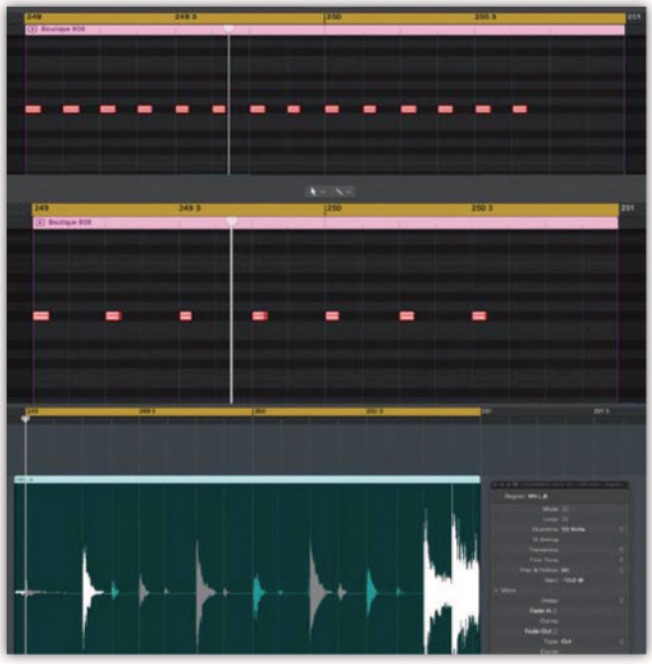
If your drums are mostly to your liking and have a great feel, you may only need to move a few hits here and there. But rather than using your eyes to determine how in-the-pocket the drums are, use your ears and try to feel what's going on. I stress this because I've fallen into the trap of attempting to lock a drum performance to the grid to the point that the feel is lost. It's definitely something to be wary of when editing drums.

Another potential pitfall is to move one errant hit to the grid without considering the overall groove. Move the hit a little at a time until it sits nicely with what's happening before and after it.

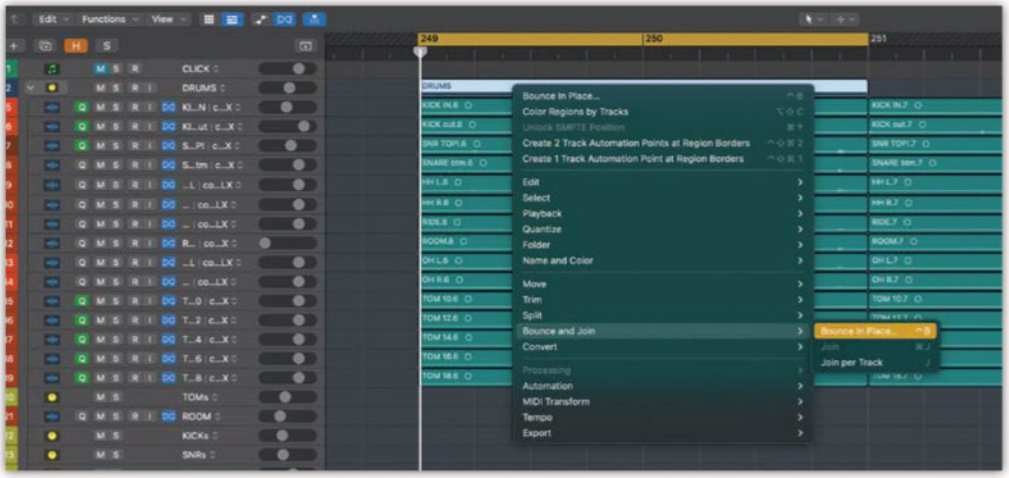
To move a transient, simply grab the marker and slide it forward or backyard in the timeline. You'll see the move updated in all of the other drum tracks. Make sure to grab the marker in the bottom half of the region to use the three-pronged option. This will allow you to move the hit while keeping the other hits on either side in the same place. The single transient option you get when click-dragging in the top half of the region moves everything in the region, which is definitely something we don't want here.

To Q Or Not To Q?

While manually nudging drum hits may seem overwhelming, this truly is the best



Logic's quantise behaviour works differently for audio and MIDI. With MIDI data, eighth notes are forced to quarter notes when quantising is set to 1/4, but with audio, only transients on or near quarter notes will be moved, while the eighth-note upbeats remain in place.



You can't create a Groove Template directly from multitrack drums, but you can easily create a stereo bounce for the purpose, by drawing an empty region on the track stack lane and bouncing in place.

way to start. But why not just quantise the whole thing?

There are two considerations to, um, consider. Firstly, any really good drummer is going to play the way they want the drums to sound. That doesn't mean they don't make minor timing errors, but the majority of what they're playing is what they think works. That could mean playing the snare a little behind the beat or getting on top of the beat in the choruses. Do you really want to undermine their artistry and professionalism by snapping every hit to the grid?

Second, is the rest of the band exactly on the grid? Probably not. There's nothing wrong with everybody being off the grid if they're doing it together. Also, keep in mind that every style and genre is going to put different demands on a drum performance. A dance track probably should be quantised while a jazz track almost certainly should not be.

Selecting your entire drum performance and applying quantisation to it is a solution that does work well in many cases. Of course, you would still want to listen and make adjustments and even play with the Q Strength value. But if your tracks are more groove-oriented and need subtle refinement, you might want to explore further.

Audio Quantisation

While Logic can quantise Flex-enabled audio in the same manner it would with MIDI, it's worth examining how Logic approaches audio quantisation. I couldn't find any explanations of this online, so I had to do a few experiments to draw these conclusions.

With MIDI quantisation, Logic will move all of the notes to the nearest division of the quantise value. If you record eighth notes and quantise to quarter notes, Logic will stack the upbeat eighth notes on top of the downbeats, leaving you with only notes playing on downbeats.

With audio, it seems, Logic does a bit of interpretation, and only moves the notes that are close to the quantise value. So If you have a hi-hat playing eighth notes and you quantise it to quarter notes, Logic will move the quarter-note downbeats to the grid, but will leave the upbeats untouched.

Understanding this concept gives you more options when quantising a drum performance. The common approach to quantising has always been to determine the smallest note value in the take and quantise to that value. Now you are free to only quantise the first beat of the bar, for example (Q=1/1 Note), while keeping the rest of the measure's feel intact. Use the Q Strength parameter to aid in making the performance even more natural.

If this nuanced approach is more than you need, your simple takeaway should be that you can still quantise a straightforward rock track to eighth notes even though there are 16-note fills. Logic won't make them eighth-note fills. It's too logical for that.

Groove Templates

Another method for quantising your live drums is to use Groove Templates. I haven't had success creating a Groove Template from multitracked drums yet, so, instead, I bounce the multitracks down to a stereo file and use that.

First, find a two-bar section of your drum take that's as close to ideal as possible. Make any adjustments with the Flex markers if needed. On your drum bus (or drum track stack), use the pencil tool to draw a region that's the length of your optimal groove. Now, right-click on that empty region and choose Bounce In Place. Set it to create a new track and to leave the original.

Now you'll have a stereo bounce on a new track. Enable Flex on that region, go to the Region Inspector, scroll down to the bottom of the Quantize menu, and choose Make Groove Template. Then select your entire drum performance (or just the sections that need help) and, in the Quantize drop-down, choose the newly created groove.

Tidy Toms

In the previous installment of this series, I described a quick and dirty method for automating the toms' level without touching the tracks themselves. Now that we have our drums fully edited, we can start getting a bit more surgical on them.

A tried-and-true method is to add the Noise Gate plug-in to each tom track. Start by adding it to the first tom, then dial in settings so that the gate only opens when the drum is hit. Make sure to go to a loud section of the song and ensure that the gate isn't opening on the rest of the kit.

After you have your settings, copy the plug-in to the remaining tom tracks by Option-dragging it, and tweak the settings as needed.

However, if the performance is simply too dynamic, Noise Gates are not going to cut it. If you've been reading my columns here for a while, you can probably predict my first suggestion: make a duplicate Track Alternative and add 'Tom Edits' to the name. You can achieve great results by simply using the Marquee tool to select the measures where the toms are inactive and then muting them (just remember to turn Groups off first).

As a final step, select all of the regions (the muted ones too) and add a small fade in/out from the Region Inspector window. It may not be obvious where to add this. You need to double-click next to the Fade In/Out drop-down menu for the field to be exposed.

Q What happens when I change a mic preamp's input impedance?

I am trying out an ART Voice Channel (https://artproaudio.com/product/voicechannel-tube-channel-strip-with-digital-outs), which has some interesting features, one of which is a dial to vary the microphone impedance continuously from 3kΩ all the way down to 120Ω. So I have been testing this with a bunch of dynamic and capacitor mics, which have output impedances ranging from 100Ω to 350Ω. I've also tried this with a Triton FetHead [*in-line booster preamp*].

I've read Hugh's 'Understanding Impedance' article (*SOS* January 2003: www.soundonsound.com/techniques/understanding-impedance) and some other sources on the topic. I also know about the '10:1 rule' and understand that dropping below that ratio can cause a high-frequency roll off, but it's typically very high so not really audible.

What I've learned so far is that while I cannot hear any frequency response change, the mic sensitivity changes a lot when I lower the input impedance. So, naturally, I then turn up the preamp gain and then the sound can change a lot — the ART Voice Channel has a tube, but lots of mic pres do nice things when driven.

There seems to be lots to discover here, but initially I have two questions. First, am I right that if I can't hear any HF roll off I should not worry about anything? Second, if the impedance lowering also lowers the sensitivity, then I use the gain on the preamp to bring it back up, am I right that I am engaging more of the preamp's natural tonal breakup colour — or am I just introducing more noise because the preamp is not being overloaded in this scenario?
Andrew Culver

ART's Voice Channel has a variable input impedance: it won't be useful on all mics, but it can affect the voicing of moving-coil dynamic mics and some others which feature output transformers.

Hugh Robjohns, Technical Editor

The idea of switchable or variable input impedance is not new. The classic Neve 1073 preamp from the '70s had provision for selecting either 300Ω or 1.2kΩ input impedances, as did many preamps of that era — often by reconfiguring the input transformer windings between series and parallel. This was because some institutions were still working in a matched-impedance environment, and others in the modern matched-voltage format. But engineers/producers found that switching impedances was sometimes musically beneficial, and in more recent years manufacturers have seized on the idea as a selling point.

To address your main question, lower input impedances will definitely reduce the mic's signal level significantly because a greater portion of the mic's output voltage develops across the mic's output impedance, and less across the preamp's input impedance. So lower input impedances will need more preamp gain to restore the preamp output level. As you say, that will inevitably increase noise. In some cases it will increase the amount of distortion too, but not because you are 'driving' the preamp with signal level (if you're just compensating for the drop in level, then the signal is no hotter than before!) but because some preamps introduce more self-distortion at higher gains — this depends on active device linearity and the amount of negative feedback. The preamp's own frequency response may also change at high gain, and when it does so it's typically a rolling off at the high end.

There may be tonal effects on the mic itself when driving a low impedance. Not just a change to brightness, but also a change to fast transients and dynamic range, potentially resulting in a thicker, more solid, denser sound. How much you hear this is obviously dependent on the source as well as the mic. Dynamic (moving-coil) mics, and mics with output transformers, tend to be the ones that respond most to impedance changes because their output impedance changes significantly with frequency. Consequently, as the mic output impedance and preamp input impedance form a potential divider, the mic's frequency response changes depending on the preamp input impedance. Often subtly, sometimes dramatically. The closer the preamp impedance is to the average mic output impedance, the more pronounced the changes will be, and the higher the input impedance the more consistent and neutral. Some vintage designs were intended to feed 30Ω or 50Ω inputs, but most expected 200Ω to 300Ω, rather than the 1.5kΩ standard we have today. So it could be argued that they only deliver the frequency response intended by the designer — and thus sound 'right' — when facing a low(ish) impedance.

I do have a couple of mic preamps with variable impedance (a Focusrite ISA 428 and an Audient ASP880) and both normally stay at their high setting. But having said that, most of my mics are transformerless capacitor mics or active ribbons. Capacitor mics with output transformers can respond noticeably to impedance variations (mainly vintage valve mics and vintage capacitor mics), but more modern transformerless designs don't respond well to low impedance loads at all: transient distortion and a reduced headroom are common indications of an unhappy mic. So it's best to keep the impedance above 1.5kΩ for them, and 2.5 to 5 kΩ is ideal. Most passive ribbons prefer much higher impedances, typically over 5kΩ, and dedicated ribbon preamps usually offer inputs at 15-30 kΩ.

It's worth noting that even where you have a mic whose character changes with impedance, the sonic differences from changing the impedance usually won't be huge. They will often be noticeable, though, and whether the result is better or worse at different impedances is really a subjective and contextual thing.

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SOS WHY I LOVE... CHANGE

EMILY LAZAR

When Catherine Marks asked me to be a part of this special issue of *SOS*, I was honoured to be included among this awesome community of women, and also grateful for an opportunity to reflect.

When I started, as an artist and songwriter, there was a distinct line between the different roles involved in making a recording — and if you were a woman, that line was considerably harder to cross. Without the jargon to properly communicate what I wanted to achieve sonically, choices were made for me. Pretty quickly I made the decision to cross over onto the other side of the glass and learn as much as I could, so I'd be in control of my own choices.

For me, making those choices is about listening to my gut. It's not calculation, it's instinct. It's not science, it's art. That's what my studio The Lodge has always been about. I wanted an environment where producers and engineers feel comfortable. A place with good food, art on the walls, instruments to pick up and play, and windows... definitely windows. I followed my instincts and took the risky choice to build it the way I envisioned it.

I never saw myself as an innovator, and I didn't set out to be an industry change maker. It all started by identifying necessary problems to solve. We were the first to provide extra rooms where last-minute overdubs and mix tweaks could happen during the session. We were the first to provide stem mastering, a powerful new technique that would change industry standards for mastering. I've learned over the years that being a great engineer/producer is to be a disruptor. Every project has different problems, and you need to make choices about how you address them. When necessary, you have to be prepared to disrupt a mix, to tinker with it until it allows the song to tell the story it needs to tell.

Remaining tuned in to my instincts led me to founding We Are Moving The Needle to address gender disparities in the industry. WAMTN amplifies the voices that have always been out there but haven't always been heard. We build community, with more than 75 industry leaders, artists, producers and engineers on our soundBoard. We published our first *Fix The Mix* report this year, shedding light on the disparities facing women and non-binary professionals in the field. I wanted to disrupt the narrative and highlight for industry decision makers that they have the power to make positive choices, and their choices matter.

It's a beautiful time for our industry right now. The technology is more powerful than ever, and the lines have blurred so that the roles are not as rigid. No two paths are the same in this industry. Each new artist, engineer and producer embarks on a personal journey filled with choices. Keep making them. Moments that feel like roadblocks may actually be doors if you let them be. There will be hard choices, weird choices, and laughable "how did I get here" choices. Trust your instincts and enjoy the ride, wrong turns, bumps and all.

W www.wearemovingtheneedle.org

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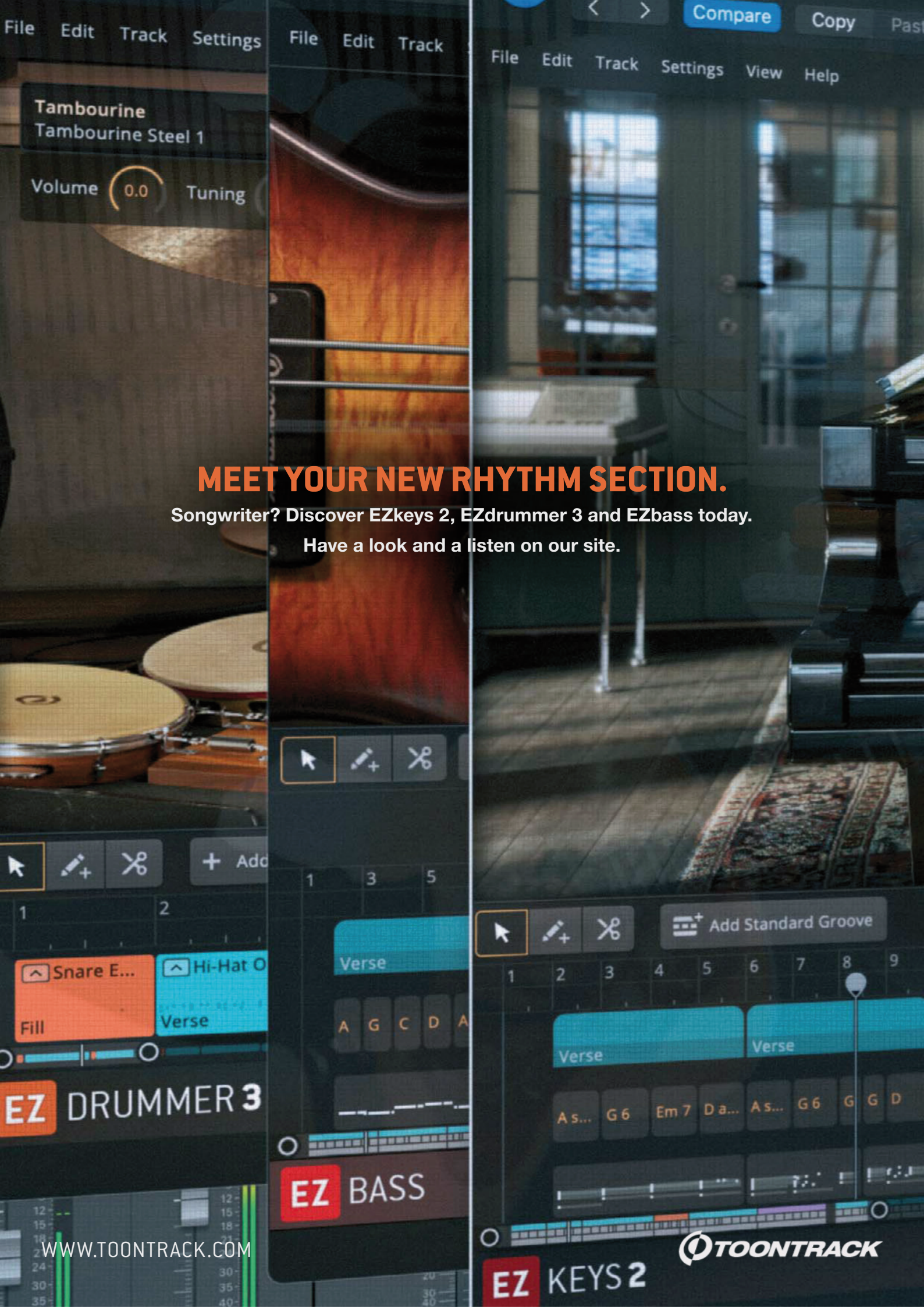

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